PROCEEDINGS

41ST ANNUAL BROADCAST ENGINEERING CONFERENCE

INCLUDING SPECIAL REPORTS:

Voluntary National NRSC Standard
Modulation, Overmodulation, and Occupied Bandwidth
Advanced Television Systems

NAB
BROADCASTERS

NATIONAL ASSOCIATION OF BROADCASTERS
DALLAS, TEXAS
1987

J. BALLARD NBC

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National Association of
NAB
BROADCASTERS

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March, 1987

Dear Reader:

These Proceedings contain many of the technical papers presented during the 41st Annual Broadcast Engineering Conference held in conjunction with the NAB Convention in Dallas, Texas March 27-31, 1987.

This volume consists of technical papers of interest to broadcast engineers and technicians in all areas of radio and television. Subjects covered include AM improvement, radio and television engineering and new technology, stereo sound for television, using satellites for gathering news, radio production, and advanced television systems.

Note that we have a special section which includes the recently adopted NRSC AM transmission standard, a landmark paper on AM modulation and over-modulation, and a tutorial on the subject of advanced television systems that describes an evolutionary approach to HDTV.

Take time to read and learn from these papers which have been prepared with great care by their authors. To a large extent the future of our industry's technical development relies on your interest and ability to understand and utilize the technological concepts and applications described here. In many respects they are the blueprints of our future and a valuable reference source to complement the Proceedings of past years and the NAB Engineering Handbook. Further, in the increasingly diverse and competitive broadcast marketplace, engineering remains an essential factor in the maintenance of high quality signal transmission and the ability of a station to compete effectively and provide the high level of service our local communities have come to expect from broadcasting.

We at NAB are proud to publish these Proceedings. Your comments on the Proceedings or any aspect of the 1987 Engineering Conference are welcome.

Best personal regards,

Thomas B. Keller

Thomas B. Keller

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BROADBANDING AM ANTENNAS FOR HIGHER FIDELITY AND STEREO

Alan W. Parnau
Capital Cities/ABC
New York, New York

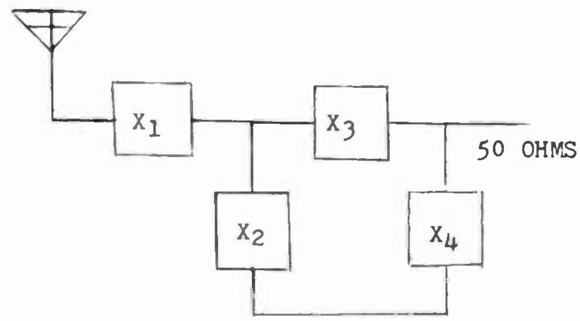
Since the advent of AM stereo in the mid 1970's, much has been written about the need for broadband antenna systems for the acceptable transmission of the AM stereo signal. While much theory has been presented, it comes to mind that an actual design example of a broadband antenna matching network might shed some light on the subject. What follows is a design example of such a network for a single tower, non-directional station.

The first step in designing a broadband matching network is to find out what you are matching to what. Since some recent construction near the base of the tower of WABC-AM in New York caused a base impedance change, I had occasion to re-sweep the base drive point. The following is the result of that sweep.

FREQUENCY (KHZ)	RESISTANCE (OHMS)	REACTANCE (OHMS)
740.5	162	-j309.3
745.0	155	-j304.7
750.5	147	-j295.8
755.0	139	-j288.7
760.5	129	-j281.4
765.0	126	-j275.8
770.0	121	-j267.5
775.0	116	-j261.9
780.5	112	-j255.0
785.0	105	-j248.4
790.5	99	-j241.6
795.0	109	-j215.1
800.5	93	-j226.1

The transmission line impedance is 50 ohms.

A diagram of the broadband network is as follows.



The job of X_1 in this system is very simple, tune out the reactance of the tower at the carrier frequency. Since the tower reactance at carrier is 267.5 ohms capacitive, we need an inductor that is 267.5 ohms at 770 Khz.

Therefore: $X_1 = 2\pi F L_1$

$$L_1 = X_1/2\pi(770{,}000)$$

$$= 267.5/2\pi(770{,}000)$$

$$= 5.529 \times 10^{-5} \text{ Henrys}$$

$$= 55.29 \text{ uH}$$

The job of X_2 is to transform the tower resistance down to a value that can then be transformed up to the characteristic impedance of the transmission line (50 ohms, in our case), by an "L" network consisting of practical components. The value of the transformed resistance we are shooting for is somewhat arbitrary at this point, it will affect things in the "L" network that follows. We should keep in mind that if the transformed resistance is too low, the components in the "L" network will not be practical. We should also remember that an "L" network cannot match equal resistances, therefore the value of the transformed resistance that we choose should not approach 50 ohms. 25 ohms appears to be a reasonable choice. There is another reason for choosing 25 ohms, which we will get into later.

The formula for X_2 is:

$$X_2=R_a\sqrt{\frac{R}{R_a-R}}$$

Where R_a is the antenna resistance at carrier, and R is the transformed value of resistance we are looking for.

$$X_2=121\sqrt{\frac{25}{121-25}}$$

$$=121(.51)$$

$$=61.75 \text{ ohms}$$

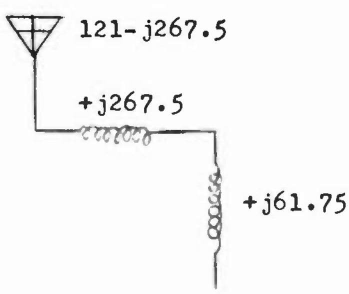
Then:

$$L_2=61.75/2\pi(770{,}000)$$

$$=1.276 \times 10^{-5} \text{ H}$$

$$=12.76 \text{ uH}$$

At carrier, our network looks like this, so far:



The impedance at the junction of X_1 and X_2 is the tower impedance in series with X_1 in parallel with X_2. The tower impedance plus X_1=121-j267.5+j267.5=121+j0. To put impedances in parallel, you calculate the product of the two impedances divided by the sum of the two impedances.

$$\frac{(121)(j61.75)}{121+j61.75}=\frac{j7471.75}{121+j61.75}$$

Now, convert both numerator and denominator to polar form. A calculator with a rectangular to polar, and a polar to rectangular button is very helpful in doing network calculations. If you don't have such a calculator, the formulas are: rectangular to polar

$$X+jY=Z\angle\theta$$

$$Z=\sqrt{X^2+Y^2} \qquad \theta=\tan^{-1}\frac{Y}{X}$$

and polar to rectangular

$$Z\angle\theta=X+jY \qquad X=Z\cos\theta \qquad Y=Z\sin\theta$$

The impedance in polar form is as follows:

$$\frac{7471.75\angle 90}{135.85\angle 27.04}$$

Now, do a polar form division. Divide the magnitude of the numerator by the magnitude of the denominator, and subtract the angle in the denominator from the angle in the numerator.

$$55.00\angle 62.96$$

and finally convert back to rectangular form.

$$25.00+j48.99$$

At 10 KHZ above carrier, or at 780 KHZ the tower impedance is 112-j255.0, from our bridge measurements.

$$X_1=2\pi FL_1$$

Remember, F is now 10 KHZ higher, so X_1 will be higher.

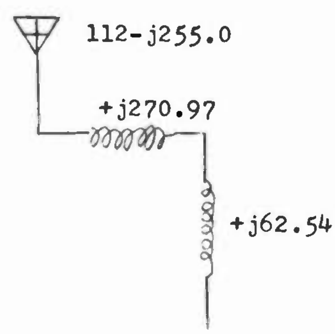
$$X_1=2\pi(780{,}000)(55.29 \times 10^{-6})$$

$$=270.97 \text{ ohms}$$

$$X_2=2\pi(780{,}000)(12.76 \times 10^{-6})$$

$$=62.54 \text{ ohms}$$

At plus 10 KHZ, our network looks like this:



The impedance at the junction of X_1 and X_2 is now:

$$(112-j255.0+j270.97)\,\|\,(j62.54)$$

$$=\frac{(112+j15.97)(j62.54)}{(112+j15.97+j62.54)}$$

$$=\frac{-(998.76-j7004.48)}{112+j78.51}$$

Converting to polar form:

$$=\frac{7075.33\angle 180-81.89}{136.78\angle 35.03}$$

Doing the division:

$$=51.73\angle 63.08$$

Converting to rectangular:

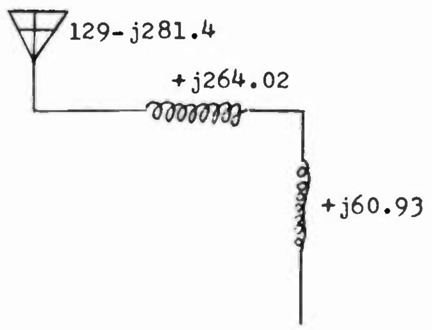
$$=23.42+j46.12$$

At 10 KHZ below carrier, or 760 KHZ:

$$Z_{tower}=129-j281.4$$

$$X_1=+j264.02$$

$$X_2=+j60.93$$



The impedance at the junction of X_1 and X_2 is:

$$\frac{(129-j281.4+j264.02)(j60.93)}{129-j281.4+j264.02+j60.93}$$

$$= \frac{1058.96+j7859.97}{129+j43.55}$$

$$= \frac{7930.99\angle 82.33}{136.15\angle 18.65}$$

$$=58.25\angle 63.68$$

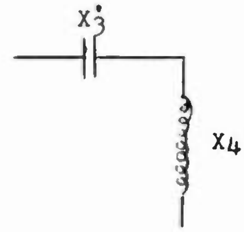
$$=25.83+j52.21$$

To summarize, we have the following three impedances at the junction of X_1 and X_2 at carrier and plus and minus 10 KHZ:

760KHZ-25.83+j52.21
770KHZ-25.00+j48.99
780KHZ-23.42+j46.12

X_3 is the series arm of an "L" network, and X_4 is the shunt arm of an "L" network. If X_3 and X_4 were single components, we would be able to provide a match to the 50+j0 transmission line at only one frequency, usually the carrier frequency. The match (or VSWR) at other (sideband) frequencies would not be as good. Another way to say this is that we would need three different "L" networks to provide a match at the carrier, upper sideband, and lower sideband. Lets now calculate the "L" networks we would need at each frequency.

Lets assume we will use a leading "L" network.



$$X_3' = -jR_2\sqrt{\frac{R_1}{R_2} - 1}$$

$$X_4 = j\frac{R_1}{\sqrt{\frac{R_1}{R_2} - 1}}$$

Where R_1 is the characteristic impedance of the transmission line, and R_2 is the resistive component of the impedance at the junction of X_1 and X_2.

At 760KHZ:

$$X_3' = -j25.83\sqrt{\frac{50}{25.83} -1}$$

$$=-j25.83(.967)$$

$$=-j24.97$$

There is also 52.21 ohms of inductive reactance at the junction of X_1 and X_2 that needs to be cancelled out. This can be accomplished by making X_3 more capacitive by 52.21 ohms. Therefore $X_3=-j24.97-j52.21=-j77.18$

$$X_4=j\frac{50}{\sqrt{\frac{50}{25.83} - 1}}$$

$$=j\frac{50}{.967}$$

$$=j51.69$$

At 770KHZ:

$$X_3'=-j25\sqrt{\frac{50}{25}-1}$$

$$=-j25$$

$$X_3=-j25-j48.99$$

$$=-j73.99$$

$$X_4=j\frac{50}{\sqrt{\frac{50}{25} -1}}$$

$$=j50$$

At 780KHZ:

$$X_3' = -j23.42\sqrt{\frac{50}{23.42} - 1}$$

$$= -j24.95$$

$$X_3 = -j24.95 - j46.12$$

$$= -j71.07$$

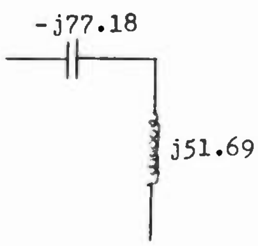
$$X_4 = j\frac{50}{\sqrt{\frac{50}{23.42} - 1}}$$

$$= j\frac{50}{1.065}$$

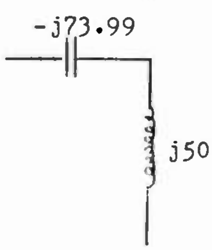
$$= j46.93$$

Our three networks are as follows:

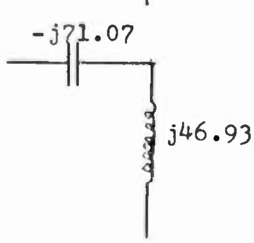
760KHZ



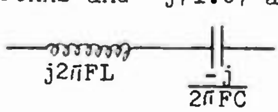
770KHZ



780KHZ

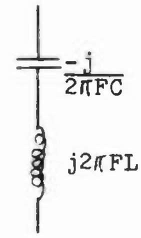


The fundamental idea in making the network broadband is not to just make X_3 a capacitor whose reactance is -j73.99 ohms at 770KHZ, and to take what you get at the sidebands, but rather to make X_3 a series combination of an inductor and a capacitor whose reactance is -j73.99 at 770KHZ, -j77.18 at 760KHZ and -j71.07 at 780KHZ.

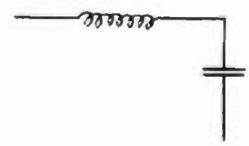


As F increases, the inductive reactance increases, and the capacitive reactance decreases, therefore the overall reactance becomes less capacitive as the frequency is increase . This is exactly what we want.

The same goes for X_4:



As F increases, the combination becomes less capacitive, or more inductive, but checking our three "L" networks, we see that we want X_4 to become less inductive with increasing frequency, therefore the leading "L" network will not give us the broadband characteristics we want. Lets try a lagging "L" network.



$$X_3' = jR_2\sqrt{\frac{R_1}{R_2} - 1}$$

$$X_4 = -j\frac{R_1}{\sqrt{\frac{R_1}{R_2} - 1}}$$

At 760KHZ:

$X_3' = j24.97$ (The same as before except the reactance is positive)

$X_3 = j24.97 - j52.21$

$= -j27.24$

$X_4 = -j51.69$ (The same as before except the reactance is negative)

At 770 KHZ:

$X_3' = j25$

$X_3 = j25 - j48.99$

$= -j23.99$

$X_4 = -j50$

At 780KHZ

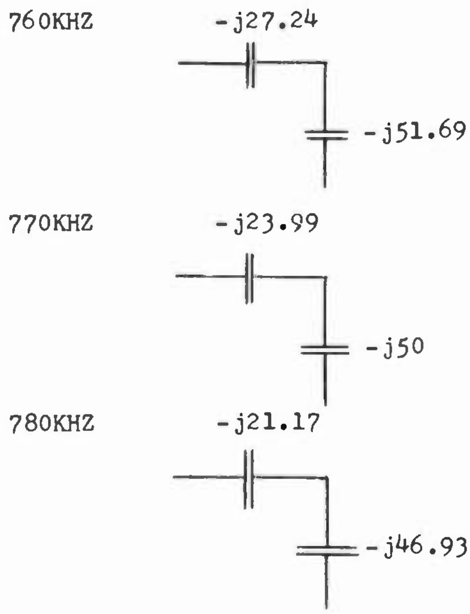
$X_3' = j24.95$

$X_3 = j24.95 - j46.12$

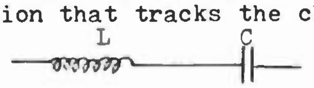
$= -j21.17$

$X_4 = -j46.93$

Our three networks are now as follows:



Since both legs now contain elements whose capacitive reactance becomes smaller with increasing frequency, we should be able to find a series L-C combination that tracks the change.



The formula for the inductor is:

$$L = \frac{(X_{lsb})(2\pi F_{lsb}) - (X_{usb})(2\pi F_{usb})}{(2\pi F_{usb})^2 - (2\pi F_{lsb})^2}$$

Where X_{lsb} is the capacitive reactance of the series arm at the lower sideband, F_{lsb} is the lower sideband frequency, X_{usb} is the capacitive reactance of the series arm at the upper sideband, and F_{usb} is the upper sideband frequency.

$$L = \frac{(27.24)(2\pi)(760{,}000) - (21.17)(2\pi)(780{,}000)}{(2\pi \cdot 780{,}000)^2 - (2\pi \cdot 760{,}000)^2}$$

$$= \frac{1.30077 \text{ X } 10^8 - 1.0375172 \text{ X } 10^8}{2.4018667 \text{ X } 10^{13} - 2.2802732 \text{ X } 10^{13}}$$

$$= \frac{.2632528 \text{ X } 10^8}{.1215935 \text{ X } 10^{13}}$$

$= 2.1650236 \text{ X } 10^5$

$= 21.65\text{uH}$

The formula for the capacitor is:

$$C = \frac{1}{(X_{lsb})(2\pi \cdot F_{lsb}) + (2\pi F_{lsb})^2 \cdot L}$$

$$= \frac{1}{27.24 \cdot 2 \cdot \pi \cdot 760{,}000 + (2 \cdot \pi \cdot 760{,}000)^2 \cdot 21.65\text{X}10^{-6}}$$

$$= \frac{1}{1.30077\text{X}10^8 + 4.9367914\text{X}10^8}$$

$$= \frac{1}{6.2375614\text{X}10^8}$$

$= 1.6031906\text{X}10^{-9}$

$= 1603\text{pF}$

If the series arm were inductive, the formulas would be:

$$L = \frac{(X_{usb})(2\pi F_{usb}) - (X_{lsb})(2\pi F_{lsb})}{(2\pi \cdot F_{usb})^2 - (2\pi \cdot F_{lsb})^2}$$

$$C = \frac{1}{(2\pi F_{lsb})^2 L - (X_{lsb})(2\pi F_{lsb})}$$

For the shunt arm:

$$L = \frac{(X_{lsb})(2\pi F_{lsb}) - (X_{usb})(2\pi F_{usb})}{(2\pi F_{usb})^2 - (2\pi F_{lsb})^2}$$

$$= \frac{(51.69)(2\pi \cdot 760{,}000) - (46.93)(2\pi \cdot 780{,}000)}{(2\pi \cdot 780{,}000)^2 - (2\pi \cdot 760{,}000)^2}$$

$$= \frac{2.4683115\text{X}10^8 - 2.2999850\text{X}10^8}{2.4018667\text{X}10^{13} - 2.2802732\text{X}10^{13}}$$

$$= \frac{.16832650\text{X}10^8}{.12159350\text{X}10^{13}}$$

$= 1.3843379\text{X}10^5$

$= 13.84\text{uH}$

$$C = \frac{1}{(X_{lsb})(2\pi \cdot F_{lsb}) + (2\pi \cdot F_{lsb})^2 \cdot L}$$

$$= \frac{1}{51.69 \cdot 2 \cdot \pi \cdot 760{,}000 + (2 \cdot \pi \cdot 760{,}000)^2 \cdot 13.84\text{X}10^{-6}}$$

$$= \frac{1}{2.4683115\text{X}10^8 + 3.1558981\text{X}10^8}$$

$$= \frac{1}{5.6242096\text{X}10^8}$$

$= 1.7780276\text{X}10^{-9}$

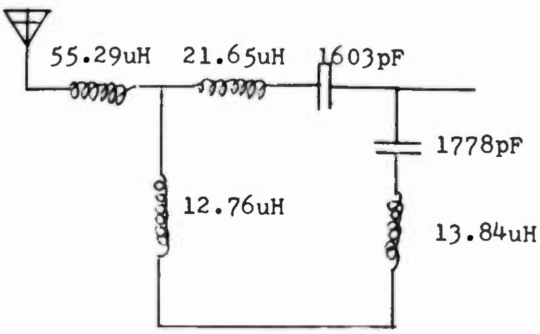
$= 1778\text{pF}$

If the shunt arm were inductive, the formulas would be:

$$L = \frac{(X_{usb})(2\pi F_{usb}) - (X_{lsb})(2\pi F_{lsb})}{(2\pi F_{usb})^2 - (2\pi F_{lsb})^2}$$

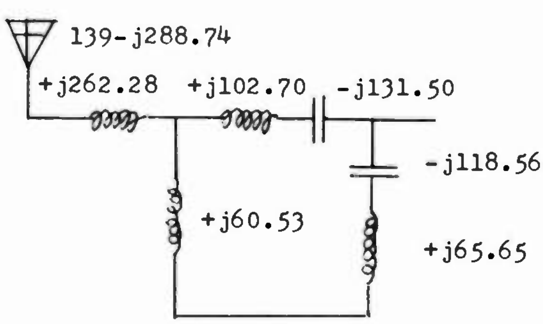
$$C = \frac{1}{(2\pi F_{lsb})^2 L - (X_{lsb})(2\pi F_{lsb})}$$

Our network now looks like this:



Since all the components are of practically realizable values, it looks like we have a good design. If this were not the case, we would have to go back and choose a new value for R, the value of the resistance at the junction of X_1 and X_2, and recalculate from there. As a check of the design, we could take the value of the tower impedance at carrier $\pm$5KHZ, $\pm$10KHZ and $\pm$15KHZ, calculate the value of the reactances of the components at each of these frequencies, and then calculate the load presented to the transmission line at each frequency. These impedances would then be plotted on a smith chart, and we can see what VSWR circle we stay within. 1.05 to 1.0 would tell us we have a good design.

At 755KHZ



$$139-j288.74+j262.28=139-j26.46$$

$$\frac{(139-j26.46)(j60.53)}{139-j26.46+j60.53}=\frac{1601.62+j8413.67}{139+j34.29}$$

$$=\frac{8564.75\angle 79.22}{143.17\angle 13.86}$$

$$=59.82\angle 65.36$$

$$=24.94+j54.37$$

$$24.94+j54.37+j102.70-j131.50=24.94+j25.57$$

$$\frac{(24.94+j25.57)(j65.65-j118.56)}{24.94+j25.57+j65.65-j118.56}$$

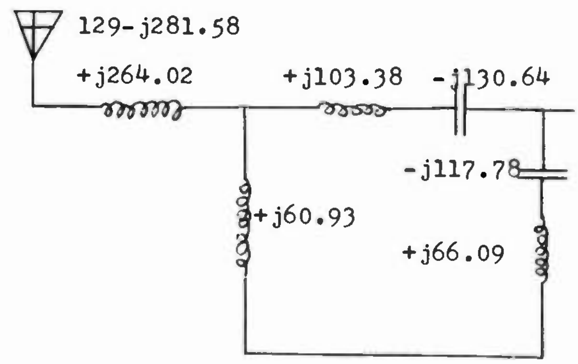
$$=\frac{1352.91-j1319.58}{24.94-j27.34}$$

$$=\frac{1889.88\angle -44.29}{37.01\angle -47.63}$$

$$=51.06\angle 3.34$$

$$=50.97+j2.97$$

At 760KHZ



$$129-j281.58+j264.02=129-j17.56$$

$$\frac{(129-j17.56)(j60.93)}{129-j17.56+j60.93}=\frac{1069.93+j7859.97}{129+j43.37}$$

$$=\frac{7932.46\angle 82.25}{136.10\angle 18.58}$$

$$=58.28\angle 63.67$$

$$=25.85+j52.23$$

$$25.85+j52.23+j103.38-j130.64=25.85+j24.97$$

$$\frac{(25.85+j24.97)(j66.09-j117.78)}{25.85+j24.97+j66.09-j117.78}$$

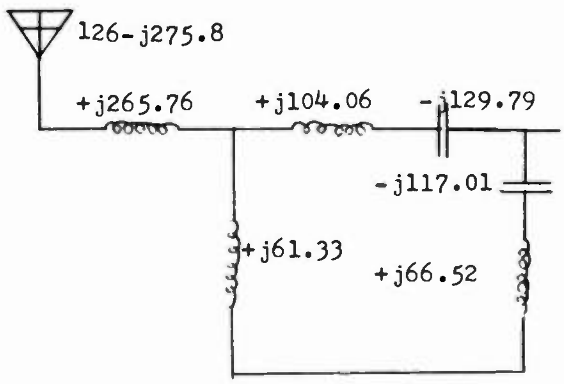
$$=\frac{1290.70-j1336.19}{25.85-j26.72}$$

$$=\frac{1857.77\angle -45.99}{37.18\angle -45.95}$$

$$=49.97\angle -.04$$

$$=49.97-j.03$$

At 765KHZ



$126-j275.8+j265.76=126-j10.04$

$$\frac{(126-j10.04)(j61.33)}{126-j10.04+j61.33}=\frac{615.75+j7727.58}{126+j51.29}$$

$$=\frac{7752.07\angle 85.44}{136.04\angle 22.15}$$

$=56.98\angle 63.29$

$=25.61+j50.90$

$25.61+j50.90+j104.06-j129.79=25.61+j25.17$

$$\frac{(25.61+j25.17)(j66.52-j117.01)}{25.61+j25.17+j66.52-j117.01}$$

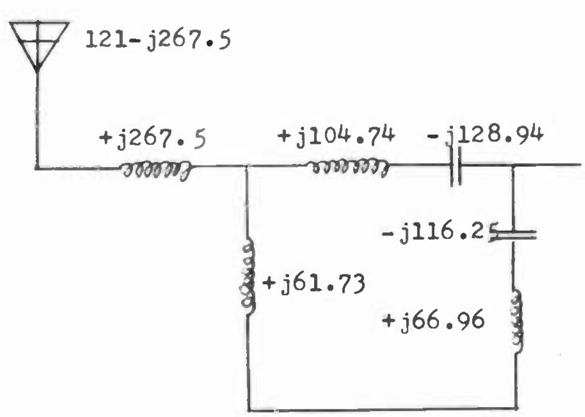
$$=\frac{1270.83-j123.05}{25.61-j25.32}$$

$$=\frac{1813.00\angle -45.50}{36.01\angle -44.67}$$

$=50.35\angle -.83$

$=50.34-j.73$

At 770KHZ



$121-j267.5+j267.5=121$

$$\frac{(121)(j61.73)}{121+j61.73}=\frac{j7469.33}{121+j61.73}$$

$$=\frac{7469.33\angle 90}{135.84\angle 27.03}$$

$=54.99\angle 62.97$

$24.99+j48.98$

$24.99+j48.98+j104.74-j128.94=24.99+j24.78$

$$\frac{(24.99+j24.78)(j66.96-j116.25)}{24.99+j24.78+j66.96-j116.25}$$

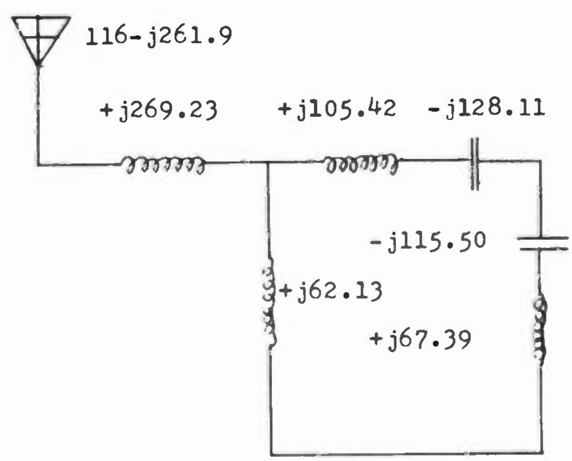
$$=\frac{1221.41-j1231.77}{24.99-j24.51}$$

$$=\frac{1734.68\angle -45.24}{35.00\angle -44.44}$$

$=49.56\angle -.80$

$=49.56-j.69$

At 775KHZ



$116-j261.9+j269.23=116+j7.33$

$$\frac{(116+j7.33)(j62.13)}{116+j7.33+j62.13}=\frac{-(455.41-j7207.08)}{116+j69.46}$$

$$=\frac{7221.45\angle 180-86.38}{135.21\angle 30.91}$$

$=53.41\angle 62.71$

$=24.49+j47.47$

$24.49+j47.47+j105.42-j128.11=24.49+j24.78$

$$\frac{(24.49+j24.78)(j67.39-j115.5)}{24.49+j24.78+j67.39-j115.5}$$

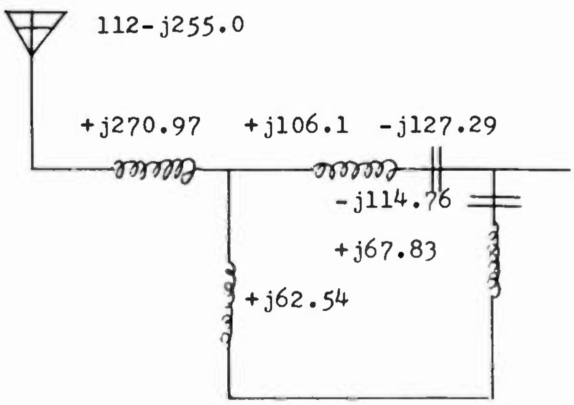
$$=\frac{1192.17-j1178.21}{24.49-j23.33}$$

$$=\frac{1676.14\angle -44.66}{33.82\angle -43.61}$$

$$=49.56\angle -1.05$$

$$=49.55-j.91$$

At 780KHZ



$$112-j255.0+j270.97=112+j15.97$$

$$\frac{(112+j15.97)(j62.54)}{112+j15.97+j62.54}=\frac{-(998.76-j7004.48)}{112+j78.51}$$

$$=\frac{7075.33\angle 180-81.89}{136.78\angle 35.03}$$

$$=51.73\angle 63.08$$

$$=23.42+j46.12$$

$$23.42+j46.12+j106.10-j127.29=23.42+j24.93$$

$$\frac{(23.42+j24.93)(j67.83-j114.76)}{23.42+j24.93+j67.83-j114.76}$$

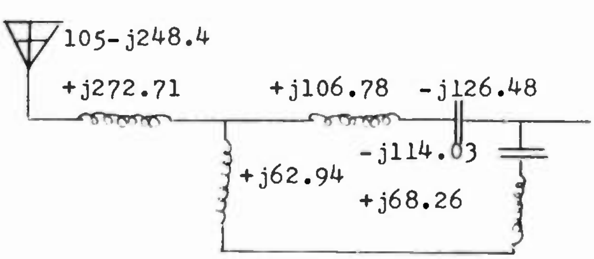
$$=\frac{1169.96-j1099.10}{23.42-j22.00}$$

$$=\frac{1605.25\angle -43.21}{32.13\angle -43.21}$$

$$=49.96\angle 0$$

$$=49.96+j0$$

At 785KHZ



$$105-j248.4+j272.71=105+j24.31$$

$$\frac{(105+j24.31)(j62.94)}{105+j62.94+j24.31}=\frac{-(1530.07-j6608.70)}{105+j87.25}$$

$$=\frac{6783.51\angle 180-76.96}{135.75\angle 39.73}$$

$$=49.97\angle 63.31$$

$$=22.44+j44.65$$

$$22.44+j44.65+j106.78-j126.48=22.44+j24.95$$

$$\frac{(22.44+j24.95)(j68.26-j114.03)}{22.44+j24.95+j68.26-j114.03}$$

$$=\frac{1141.96-j1027.08}{22.44-j20.82}$$

$$=\frac{1517.53\angle -41.96}{30.61\angle -42.86}$$

$$=49.58\angle .90$$

$$=49.57+j.78$$

To summarize, the load presented to the transmission line is as follows:

755	50.97+j2.97
760	49.97-j.03
765	50.34-j.73
770	49.56-j.69
775	49.55-j.91
780	49.96+j0
785	49.57+j.78

Normalizing to 50 ohms

755	1.019+.059
760	.999+.001
765	1.007-.015
770	.991-.014
775	.991-.018
780	.999+0
785	.991+.016

All frequencies are within the 1.05 VSWR circle except 755KHZ which is about 1.06. This is quite good, considering what you are used to seeing with the standard "L" "T" and "Pi" networks.

Now that the component values have been determined, voltage and currents in the various components can be determined in the normal Manner, so that the actual parts could be ordered. Silver plated coils and glass vacuum variable capacitors should be used to keep losses in the network to an absolute minimum.

One last point, if you check the resonant frequency of the shunt leg of the "L" network, you find it to be 1014580 Hz. This forms a "trap" to keep WINS, 1010KHZ out of WABC's transmitter. This frequency can be adjusted by changing the value of "R".

AN ECONOMICAL DIRECTIONAL ANTENNA FOR AM STATIONS

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An existing non-directional broadcast site can be modified to produce a directional gain of three dB, equivalent to doubling transmitter power in the direction of maximum gain, without adding another tower. This is accomplished by using one of the guy wires as a parasitic element. All insulators on the selected top-level guy are shorted except the top and bottom ones. This guy wire can then be tuned at its base.

If the tower is near 90 degrees in height, the guy requires a capacitive tuning reactance for both reflector and director performance. More capacitive reactance is required to produce a director. When the guy is tuned as a director, driver resistance is lower, and bandwidth is narrower. Thus best overall results are usually obtained by tuning the guy as a reflector when the full length of the guy wire is in circuit.

Ordinarily one might expect the reflector to require an inductive reactance at its base, since an inductor makes a wire look longer, and a capacitor makes a wire look shorter. Normally a reflector is physically longer than the driven element, and a director is shorter. Keep in mind that a guy wire is typically 12 to 15 percent longer than its tower projection. Thus a full-length top-level guy wire on a quarter-wave tower may behave as a reflector when shorted at its base, depending on how much of the tower top is cantilevered. If the tower were only 70 degrees tall, then an inductive reactance would indeed be required to make one of its top-level guys perform as a reflector.

This raises the possibility of tuning the guy by adjusting its active length. That is, why not short the bottom guy insulator to ground, then short just enough of the upper insulators to produce the desired pattern? This eliminates the need for a tuning reactance alltogether. Figure 1 shows two of the many patterns which can be obtained in this way when the tower is a quarter wave tall. Bandwidth is also best when no tuning reactance is used.

It may sometimes be convenient to drive the guy-wire, and tune the tower. Since the tower is not as long as the guy, it requires somewhat less capacitive tuning reactance at its base. Comparing Figures 2 and 3, where identical tower and guy dimensions are used, one can see that similar gains are obtainable. However, the input impedance of the driven tower case (Figure 3) is about half that of the driven guy case (Figure 2). This is not too important, as the bandwidths of the two configurations are comparable. However, one case may be easier to match to the transmission line impedance. As expected with this close element spacing, bandwidth is rather narrow compared to a non-directional tower alone (see table of impedances in Figure 3).

Figure 4 compares the vertical patterns of a 250 foot non-directional tower to that of the driven-tower, tuned-guy arrangement of Figure 3. Note the significant increase in high-angle radiation contributed by the parasitic guy wire. This may affect the contours of the night-time fading zone, but that is very dependent on the specific ground conductivity of the area in question.

Allow me to point out that very-high-angle radiation is not likely to be refracted back to earth by the ionosphere, and even if it were, the return signal would be too weak to affect communication in the primary service area. For example, Figure 3 shows a field of 109 mV/m at a mile straight up. The E layer of the ionosphere is about 60 miles up at night, making a round trip of about 120 miles. Even if the straight-up signal were perfectly reflected, the returning signal would be less than one mV/m at the ground.

A horizontally polarized field component exists for elevation angles outside of the tower/guy or the azimuth planes (Figure 5). Note that both the E_φ and the E_θ spherical-coordinate field components are parallel to the azimuth plane when the elevation angle is 90 degrees (straight up). One's sense of up, down, and sideways can become a bit disoriented in a spherical coordinate system where V-pol and H-pol are relative to the observer, not to the azimuth plane.

At any rate, calculation of the fading zone is a relatively straightforward process, and should be part of any application of this hot-guy concept of antenna design.

If desired, tuning can be accomplished with an inductor at the base of the parasitic element, rather than a capacitor, if that element is made short enough. Figure 6 employs 156 feet of hot guy wire, which can be tuned as either a director or a reflector. Note that the transition between director and reflector operation is rapid. From the standpoints of pattern bandwidth and stability, it would be best to tune the parasitic tower to the conservative side of maximum gain, away from the crossover point.

If we define the crossover point between director and reflector operation as the point where equal forward and reverse gains are obtained, some interesting correlations can be observed. Referring to Figure 7, one can see that the relative phase of the tower currents passes through 180 degrees at the crossover point. This is useful knowledge when an antenna monitor is part of the system. Another obvious feature is the peaking of base currents at crossover. This can be a useful tuning aid when an antenna monitor is not available, but an RF ammeter is at hand.

As expected, driving-point impedance changes most rapidly when tuning approaches the crossover point (Figure 8 shows the tower base impedance for the configuration of Figures 6 and 7). Since the tower currents peak at crossover, the base resistance reaches a minimum value. If an impedance bridge is available, crossover can be determined by tuning for minimum feedpoint resistance.

Some special considerations are created when one chooses to use one or more of the guy wires as array elements. First, the voltage stresses across the remaining guy insulators are usually increased, and the voltage gradient on the guy wire is also increased. Of course, the currents in the hot guys are increased. These parameters are easily calculated with general moment-method algorithms, and do need to be taken into account during the design process.

Second, some consideration must be given to improving the ground system near the base of the hot guy wire. Since the guy is acting as a second tower, its ground system should be similar to that of a normal tower. However, in light of the saving in real-estate and tower costs, this is a minor annoyance.

Third, in some installations, the bottom guy insulator may not be very close to the ground. In this case, a drop wire will have to be added if the guy is to be tuned at the base with a reactance.

Although I have not specifically shown any tall tower appplications, there is no reason parasitic guys cannot produce similar results for any height of tower.

All data were obtained using the moment method of antenna analysis.

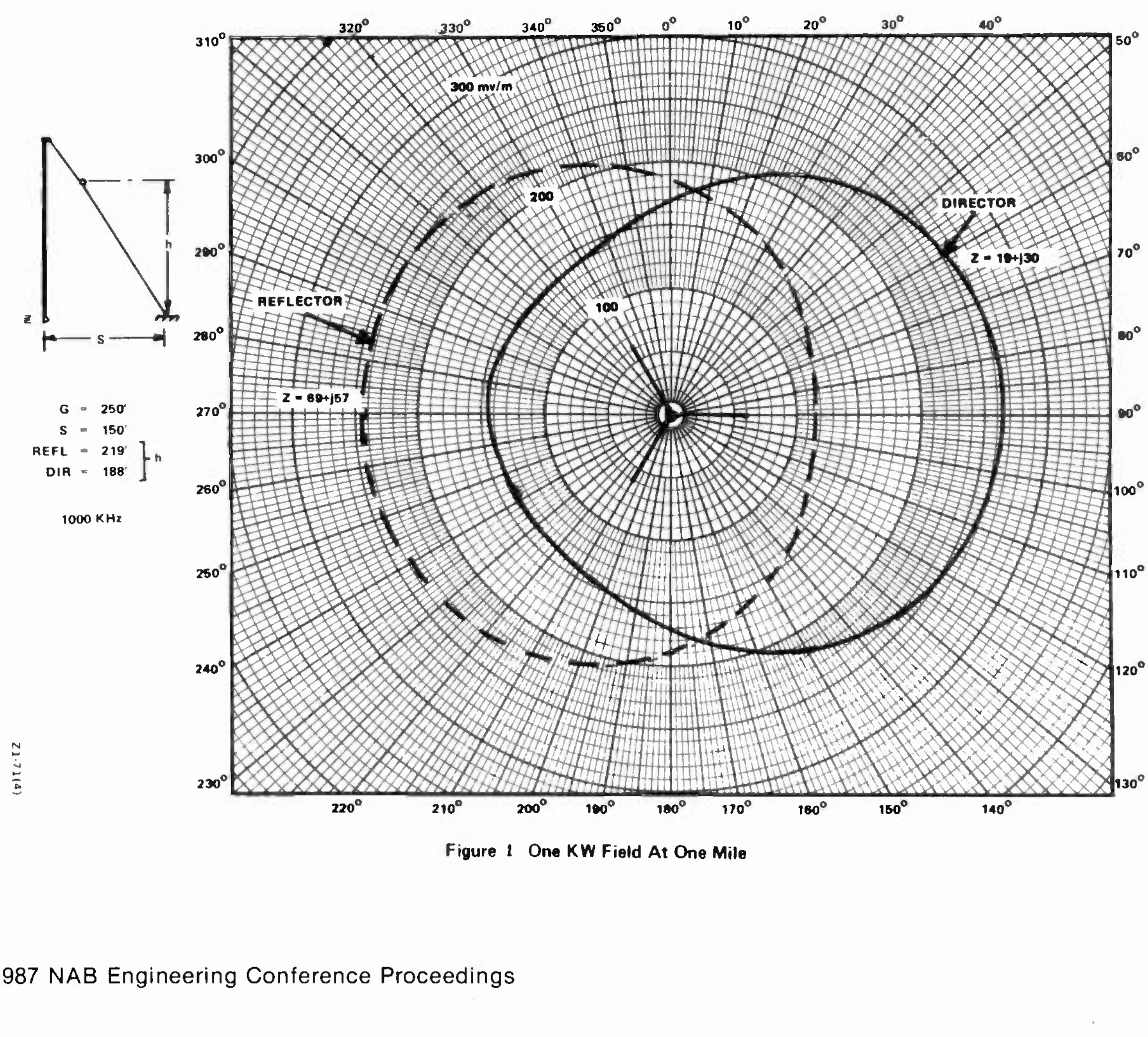
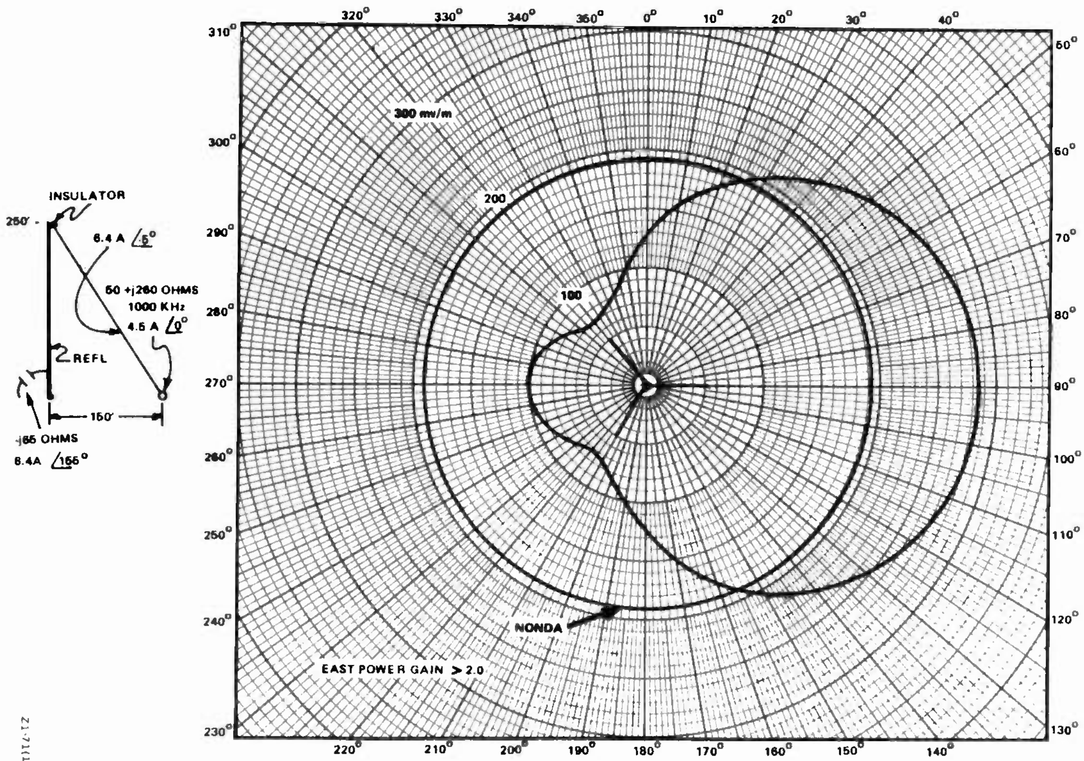
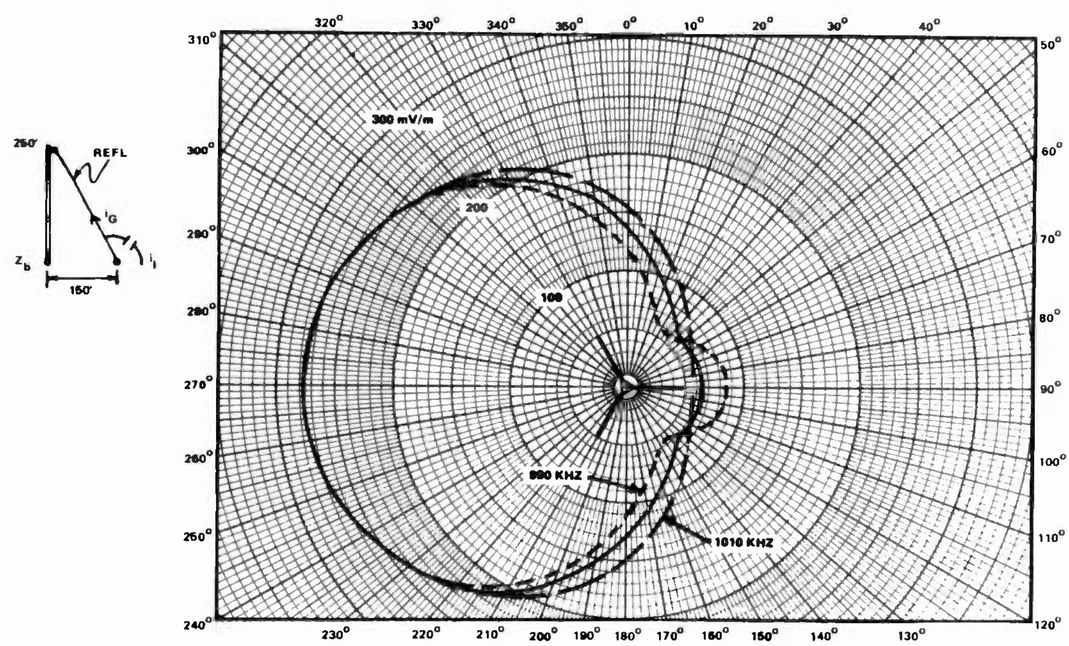


Figure 1 One KW Field At One Mile



Z1-71(1)

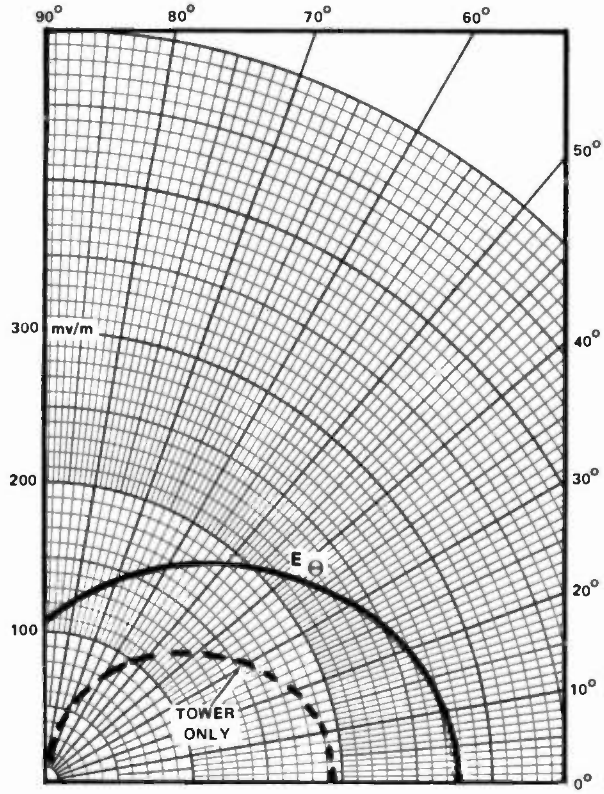
Figure 2 One KW Field At One Mile



KHZ	Z_b	Z_I	I_b	I_I	I_G
990	26.8 +j112.6	0-j161.6	6.1∠0°	6.6∠157°	8.0 ∠157°
1000	37.8 +j125.9	0-j160	5.1∠0°	6.0∠150°	7.3∠150°
1010	54.5 +j137.3	0-j158.4	4.3∠0°	5.3∠140°	6.4 ∠140°

Z1-71(3)

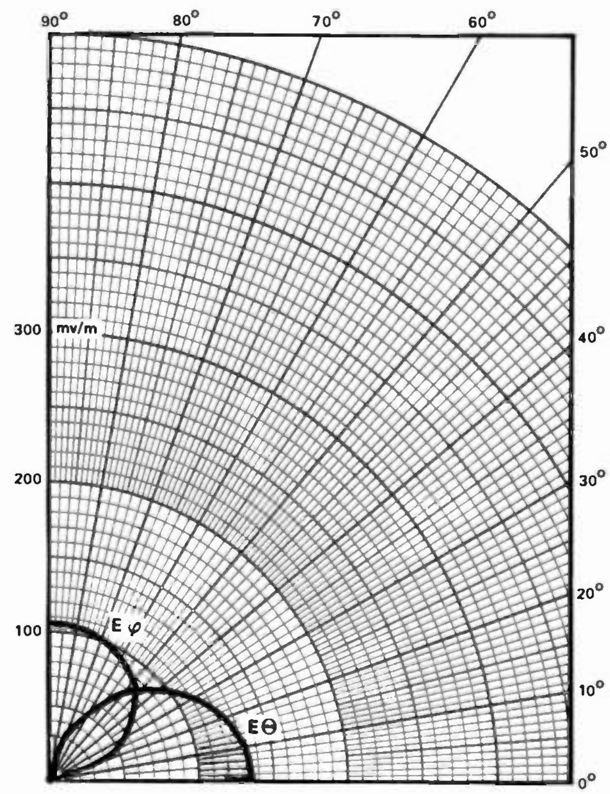
Figure 3 One KW Field At One Mile



AZ = 270°

Figure 4

VERTICAL PATTERN, 1KW, 1 MI
1000 KHz, Z_L = -j160 OHMS



AZ = 0° & 180°

Figure 5

VERTICAL PATTERN, 1KW, 1 MI
1000 KHz, Z_L = -j160 OHMS

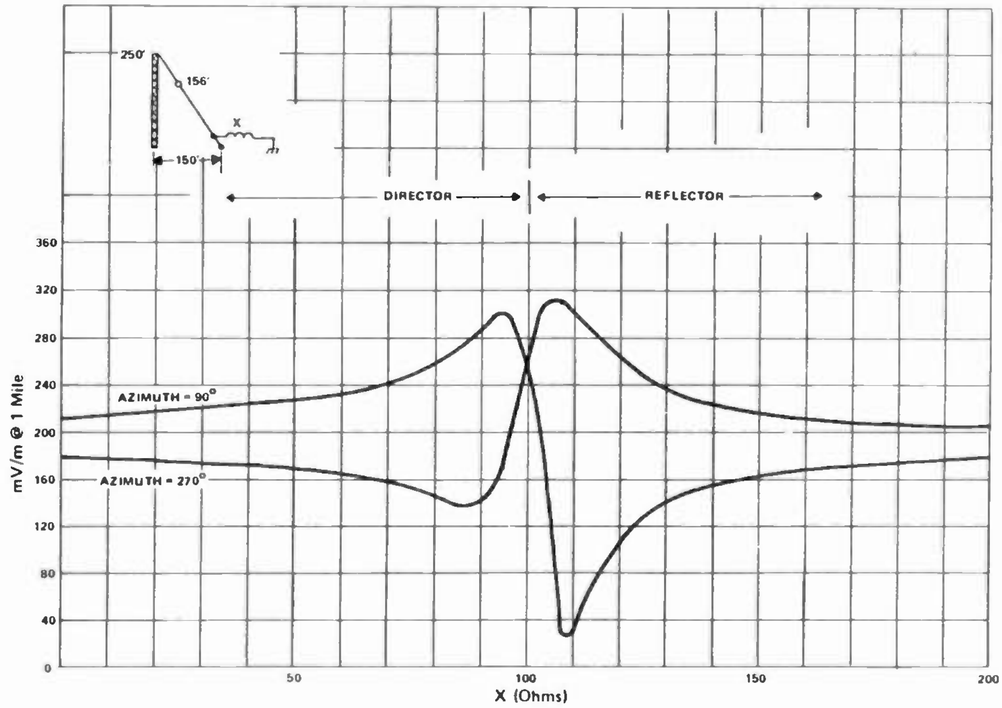


Figure 6 Field Intensity VS Tuning Reactance, 1000 KHz, 1kW

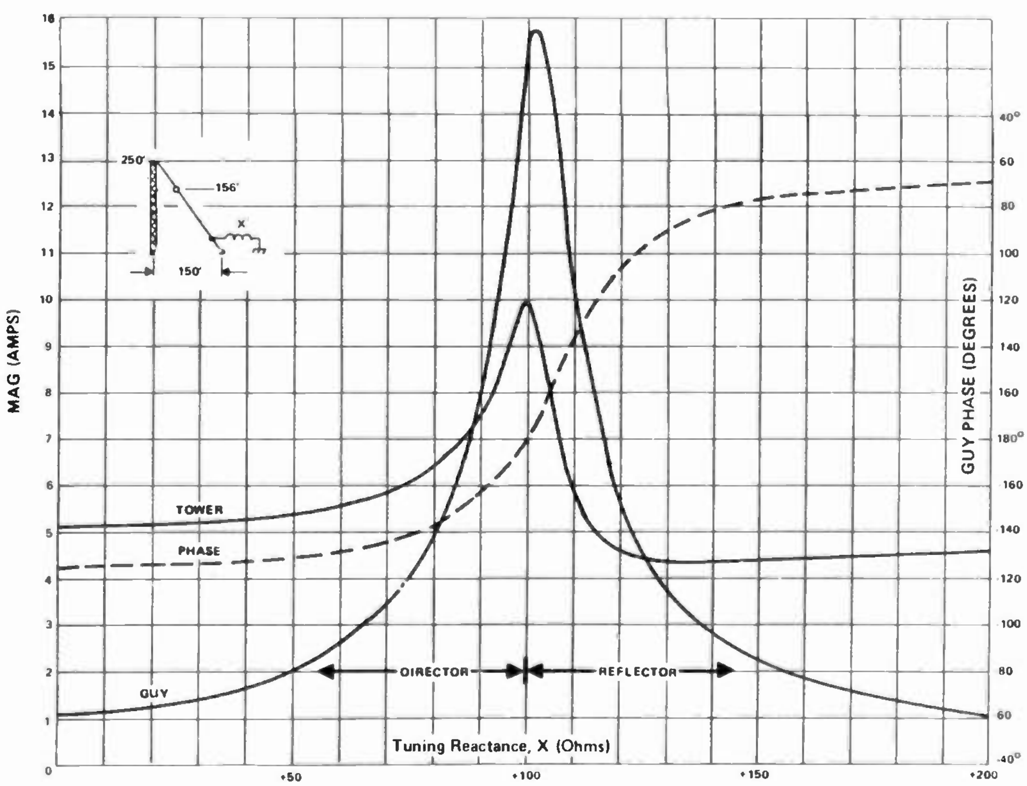
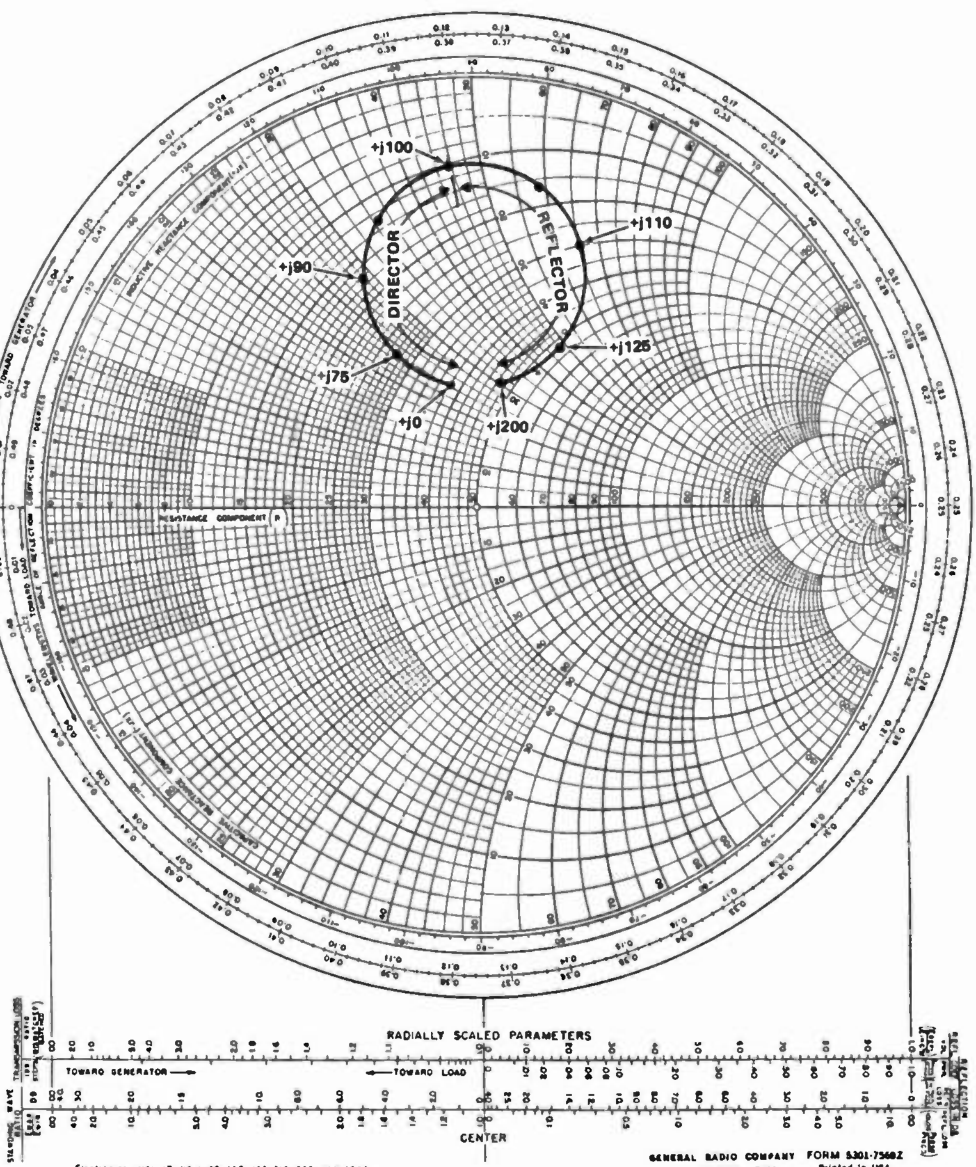


Figure 7. Base Currents VS Tuning Reactance, 1000 KHz, 1kW

IMPEDANCE COORDINATES—50-OHM CHARACTERISTIC IMPEDANCE



DATE 11-5-86

Figure 8 Tower BAse Z VS Tuning Reactance, 1000 KHz

DIGITAL AMPLITUDE MODULATION

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Harris Corporation, Broadcast Division
Quincy, Illinois

ABSTRACT

Digital Amplitude Modulation is a new medium wave amplitude modulation technique utilizing the most up-to-date data conversion systems. One of these, developed during this research, is a power multiplying digital-to-analog converter (DAC) capable of over 10 kilowatts of carrier power. This power multiplying DAC, along with a digital modulation encoder, comprises the heart of the system. The result is an AM transmitter with remarkable audio performance and overall efficiency.

This paper will also discuss how the Harris Digital Amplitude Modulator has been incorporated into a production AM transmitter.

INTRODUCTION

The Digital AM system consists of a 12 bit analog to digital converter (ADC); a digital modulation encoder, and a power multiplying DAC. The encoder converts the 12 bit digital audio code into one which is optimum for the power DAC. The digital or finite analog output of the DAC is multiplied by the radio frequency (RF) waveform to form an amplitude modulated carrier with a digital envelope. The digital AM waveform is then filtered by a bandpass output network to remove the unwanted spectral components.

The power multiplying DAC is implemented by an array of solid state Class D amplifiers. The on/off status of each amplifier is controlled by the 12 bit digital audio signal. The control signals turn on the proper number of RF amplifiers such that their sum is the desired RF output signal.

The performance of the transmitter indicates that Digital Amplitude Modulation holds great promise. For example, in a 10 KW version the transmitter's overall efficiency is 80% (60 Hz AC in to RF power out). The audio performance is equally impressive. Total harmonic distortion is less than 1.0% from 30 Hz to 12.5 kHz at 95% modulation. SMPTE 4:1 intermodulation distortion is less than 1.0% at 95% modulation. The superb phase linearity of the system is evident from the square wave response. There is essentially 0% overshoot at all frequencies. Attention has been given to stereo performance, with incidental quadrature modulation at better than -30 dB. A broadband IPA/drive design has also incorporated to accommodate the phase modulated (L-R) information.

THE DIGITAL AMPLITUDE MODULATION SYSTEM

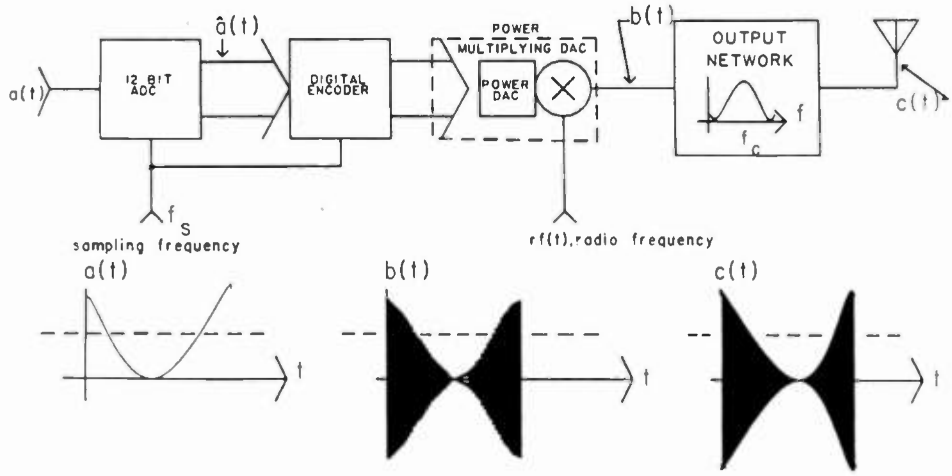
Digital Amplitude Modulation is a method of producing AM by digital means. The input audio waveform is converted to a series of 12 bit digital words by a high speed analog-to-digital converter (ADC). The 12 bit binary words are then encoded by a read-only-memory (ROM) into a form which is optimum for switching the power amplifier circuitry.

Depending on the encoder output during each sample clock cycle, the proper number of solid state Class D RF power amplifiers are turned on. The output of each amplifier is summed together to produce the desired RF level. The system block diagram along with some key waveforms are shown in Figure 1. The sampling frequency, f_s, used in Digital AM is many times the input audio bandwidth.

THE MAJOR COMPONENTS

The major components of the Digital Amplitude Modulation system are the A/D converter, the digital modulation encoder and the power multiplying digital-to-analog converter. The digital modulation encoder accepts the 12 bit digital audio signal from the ADC and encodes this into a representation which is optimum for the power multiplying DAC. The DAC is composed of an array of controllable amplifiers; their sum is the final output signal. See Figure 2.

DIGITAL AM MODULATOR



SYSTEM BLOCK DIAGRAM
AND
KEY TIME DOMAIN WAVEFORMS

Figure 1

The encoder outputs control the on/off status of each power amplifier in the DAC array. The encoder is required because the 10 kilowatt DAC is broken up into six binary weighted amplifiers and 44 equal voltage amplifiers (i.e. 2879 individual steps). The encoder derives the control signals for the 44 equal voltage amplifiers by means of a look-up table which is stored in a read-only memory. The memory is addressed by the six most significant bits of the 12 bit digital audio signal. The data stored in the encoder memory determines the proper number of equal voltage amplifiers to be turned on. The binary weighted amplifiers are controlled directly by the six least significant bits of the 12 bit digital audio signal. Every sample rate clock cycle the voltage output of each amplifier is summed together to produce the transmitter output. A block diagram of this system is shown in Figure 3, and a typical RF envelope (before filtering) showing the breakdown of

MODULATOR BLOCK DIAGRAM

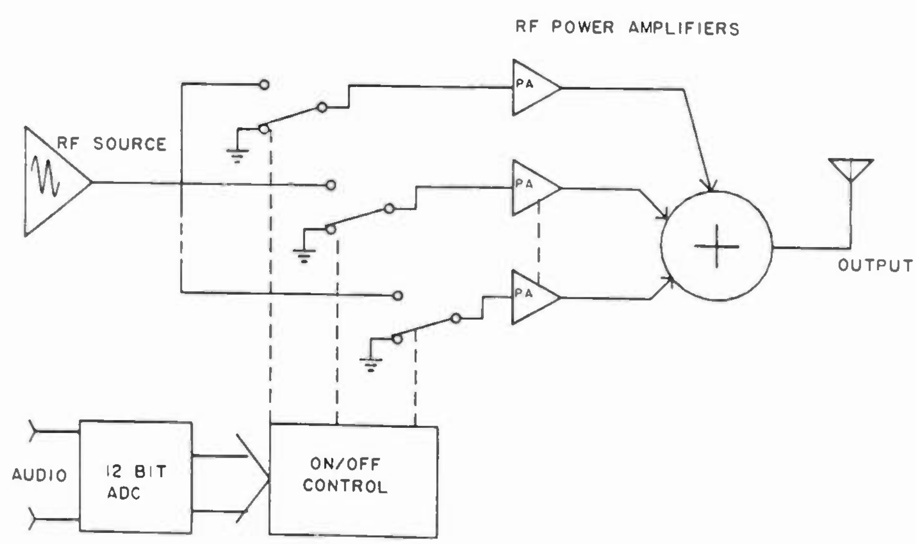
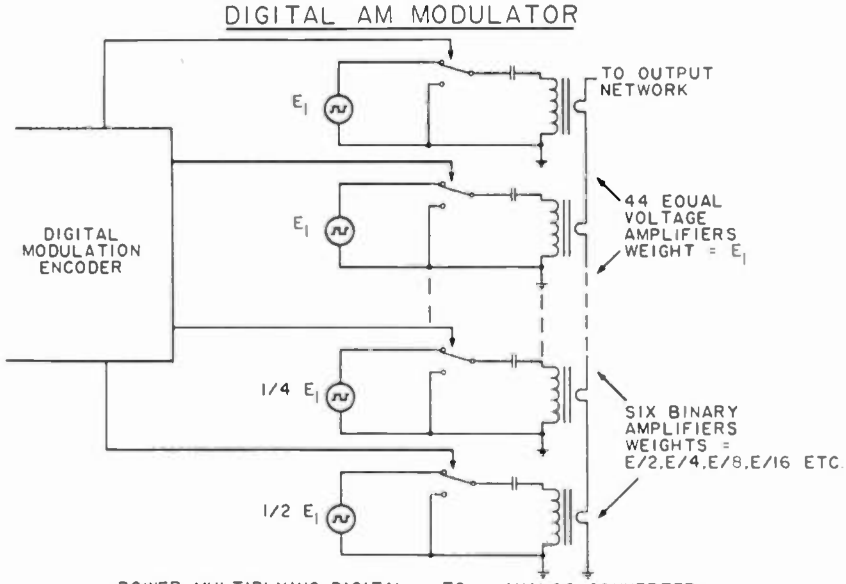


Figure 2

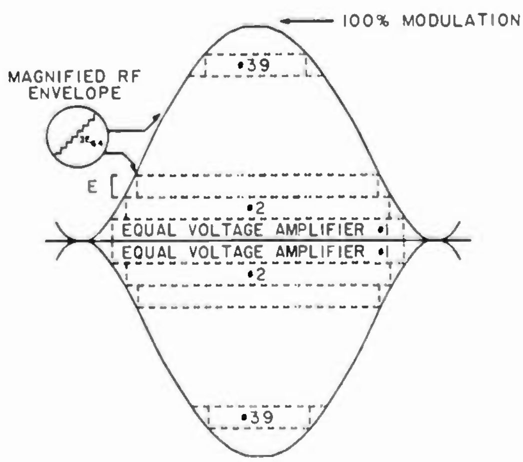


POWER MULTIPLYING DIGITAL - TO - ANALOG CONVERTER

Figure 3

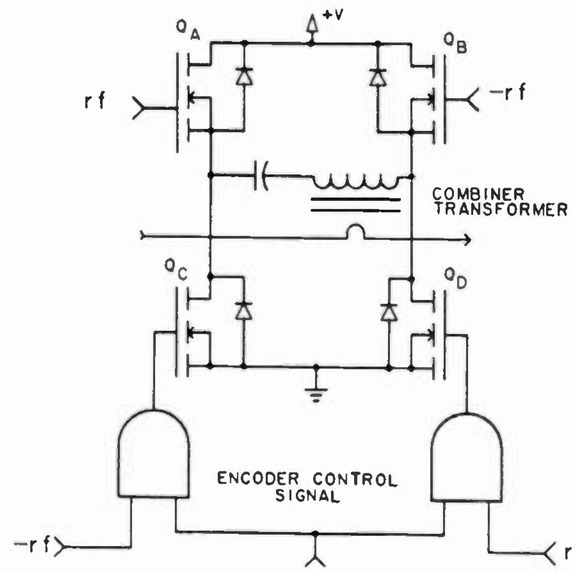
equal voltage and binary steps is shown in Figure 4.

Each amplifier of the power DAC is a solid state switch mode amplifier. The circuit is also referred to as a bridged, class D amplifier and is shown in Figure 5. If the control signal is in the logic 1 state, RF(t) is allowed to excite MOSFETs Q_C and Q_D and a voltage is developed across the load in the following manner. On the first half cycle of the radio frequency waveform, RF(t), transistors Q_A and Q_D are turned on which places a voltage, V, across the transformer. On the next half cycle, transistors Q_B and Q_C turn on and develop an equal but opposite voltage across the transformer. While in this mode, the amplifier is contributing a specific RF voltage to the combined output. If the control signal is in the logic 0 state, then the amplifier is incapable of delivering any



EXAMPLE RF ENVELOPE SHOWING RELATIONSHIP BETWEEN (FINE RESOLUTION) AND LARGE EQUAL VOLTAGE AMPLIFIERS

Figure 4



SINGLE RF SWITCH MODE POWER AMPLIFIER

Figure 5

power. The primary circuit of the transformer is, however, still complete. The drive signal for the top two devices is still present. Current is allowed to flow through the active transistor, the transformer winding, and the reverse diode of the other top transistor. This action prevents the combiner from being open circuited when an amplifier is turned off.

PERFORMANCE DATA

A 10 kilowatt Digital AM transmitter has been constructed and tested in the engineering laboratory at Harris Broadcast Division. The results indicate that this system offers substantial improvements over present modulation systems in efficiency, audio performance, and reliability.

The power amplifier is an ultra high efficiency, switch mode design. It can deliver over 10 kW of carrier power with an efficiency factor (dc input power to RF output power) of 90% or better. The audio performance is not sacrificed to achieve high efficiency, however. Digital AM can deliver a total harmonic distortion of less than 0.5% from 30 Hz to 12.5% kHz at 95% modulation. Intermodulation distortion, SMPTE 4:1, is less than 1.0% and SMPTE 1:1 IMD is less than 0.8% at 95% modulation. Another big plus for Digital AM is its square wave performance. There is essentially 0% overshoot at all frequencies which is evidence of the system's excellent phase linearity. The system also has 125% positive peak program modulation capability up to 11 kW carrier power.

When solid state devices are used in a power amplifier stage, the question of reliability always arises - particularly when extreme loads are reflected back to the power amplifier under a high Voltage Standing Wave Ratio (VSWR) condition. The Digital AM system is ideal for fast, reliable VSWR protection. When a VSWR is detected, the entire power amplifier array can be turned off by simultaneously deactivating all of the encoder output control lines. The entire turn-off process is done with low level, high speed logic circuits for minimum propagation delay. Recall that in this system, there is no modulation filter energy to dissipate as in other solid state systems. The propagation time from a VSWR detection to turning off the gate signals of the power MOSFETs is approximately 600 ns. This is much faster than previous solid state AM systems, and should enhance the transmitter's reliability.

THE HARRIS DX-10 TEN KILOWATT AM BROADCAST TRANSMITTER

A 10 kW transmitter employing this new modulation scheme has been in operation at Harris for several years and a 10 kW transmitter, the DX-10, is on display at the Harris exhibit. There are a total of 213 low cost MOSFET transistors operating at very low stress levels in the DX-10. The life cycle costs for this transmitter should be very low. Table 1 shows the performance of the DX-10.

DX-10 PERFORMANCE 710 KHz

Input:	Power	11.8 KW	6.2W
	Volts	234 V	233 V
	AMPS AØ	31 A	17.5 A
	BØ	30 A	17 A
	CØ	29 A	14 A
Power Factor		.98	.97
Calometric		9926.4 W	4963.2 W
Panel Meter		10 KW	5 KW
Supply I		53 A	27 A
Supply V		224 V	224 V
Overall Eff		84%	80%
PA Eff		87%	88%

Freq.	Dist.	Resp.	Dist.	Resp.
30	.23%	0 dB	.26%	0 dB
100	.26	0	.26	0
500	.29	0	.26	0
1K	.28	0	.29	0
2.5K	.23	-.1	.32	- .1
5K	.27	-.2	.38	- .2
7.5K	.25	-.4	.37	- .4
10K	.20	-.6	.33	- .7
IM	0.65%		0.62%	
NOISE	60 dB		60 dB	
IQM	35 dB			

TABLE 1

It should be noted that all the high powered components in this transmitter are used for generating RF power. There are no power components employed to generate high-level audio or for modulating the transmitter with associated power losses. The transmitter power circuitry, therefore, becomes very functional, because the purpose of a transmitter is to generate RF power, not audio power.

Eliminating the high level modulator in this transmitter does not alter the efficiency of the Class D RF amplifiers. In some systems where the high level modulator is eliminated, the PA efficiency is reduced and power losses in the PA increase.

RELIABILITY OF DX-10

Since the time when Harris first started working with power MOSFETS back in 1979, the cost of MOSFETS has decreased by a factor of about 10, and the ruggedness has increased by about 100 times (as measured in terms of avalanche energy capability). Therefore, the cost and reliability of the DX-10 solid state transmitters should be very favorable.

Since GE "avalanche rated" MOSFETS were installed about a year ago, there have been very few failures in this transmitter, even though the transmitter has been deliberately abused.

Several months ago, we reviewed the DX-10 with key station engineers. After the review, we decided to change the mechanical construction, primarily in the RF power amplifier section, to improve access for maintenance and ease of repair. Our engineering model is on the floor. The engineering model has four RF amplifiers on one module. If one amplifier fails, the entire module must be removed. Production model transmitters have each amplifier on a separate plug-in printed circuit card, making replacement of a module very easy and inexpensive in case of a component failure.

When an amplifier fails, an LED on the front panel changes from green to red. An output is also available at the external interface terminal board so that this fault signal can be sent to a remote location. There are also indicators on each RF amplifier module to aid in fault location.

Failure of an RF amplifier only results in a slight change in power and distortion. However, the effects of a failed amplifier can be eliminated, even with the transmitter operating, by rerouting the digital drive signal to another amplifier module. The failed amplifier can be "switched" with another amplifier normally active only on modulation peaks well beyond +100%, until a normal off-air or maintenance period occurs.

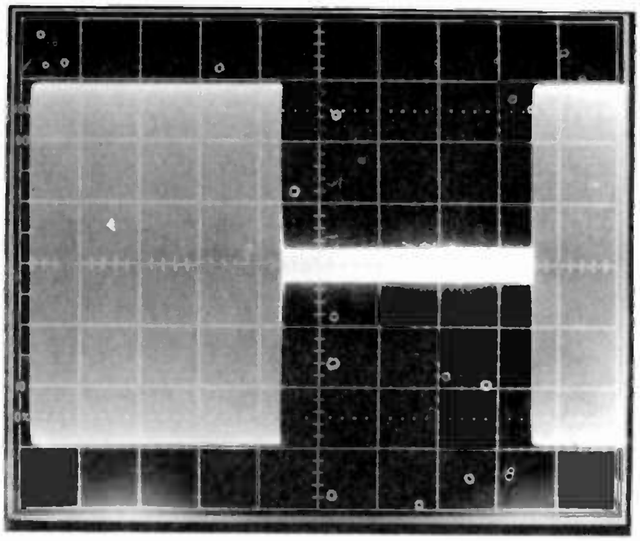
PERFORMANCE

Tuning of this transmitter is no different from tuning any high-level modulated transmitter. Loading is adjusted to obtain the correct supply current. Tuning is adjusted for maximum power output. This transmitter has inherently low distortion and thus requires no feedback.

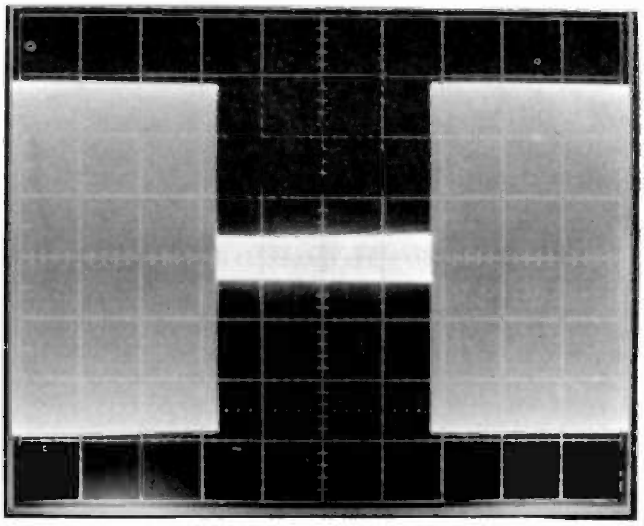
MEASURED PERFORMANCE OF DX 10

Figures 6, 7, 8, and 9 illustrate the excellent square wave response of the DX 10 at 20 Hz, 100 Hz, and 5 kHz. Figure 9 shows what happens if a complete RF amplifier fails. Distortion becomes about 2%. (Usually only one half of an amplifier will fail and will cause about 1% distortion.) Figure 10 shows what happens if 2 amplifiers fail. The distortion becomes 2.2%. Figure 11 shows the transmitter being modulated by a 100 Hz tone with all the binary weighted amplifiers disabled.

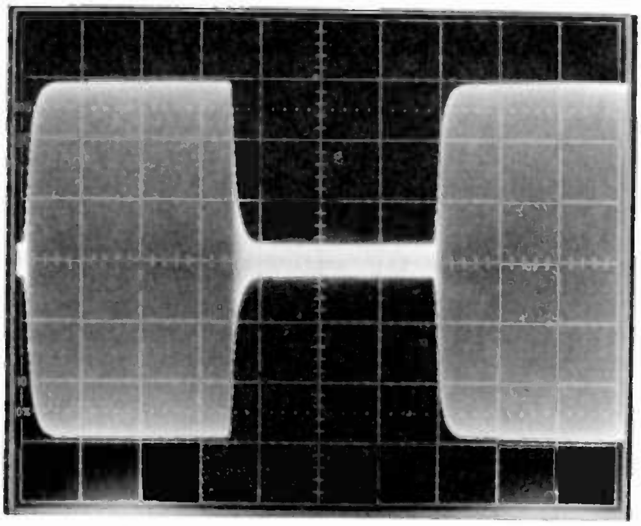
Figure 12 shows the transmitter being modulated about 100% with a 100 Hz tone. It should be noted that the transmitter is very linear even down to 100% negative peak of modulation. Figure 13 illustrates the good linearity beyond 100% positive peak modulation.



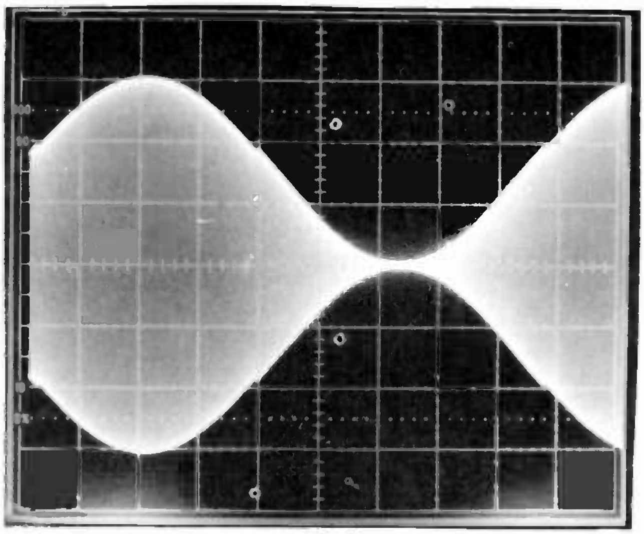
20 Hz
Figure 7



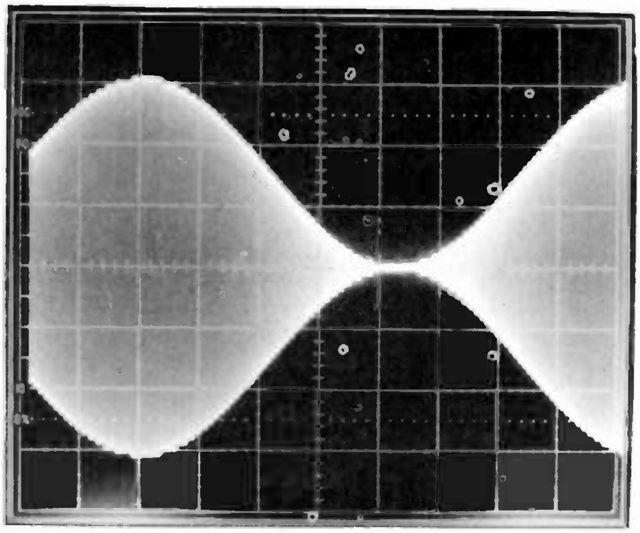
100 Hz
Figure 6



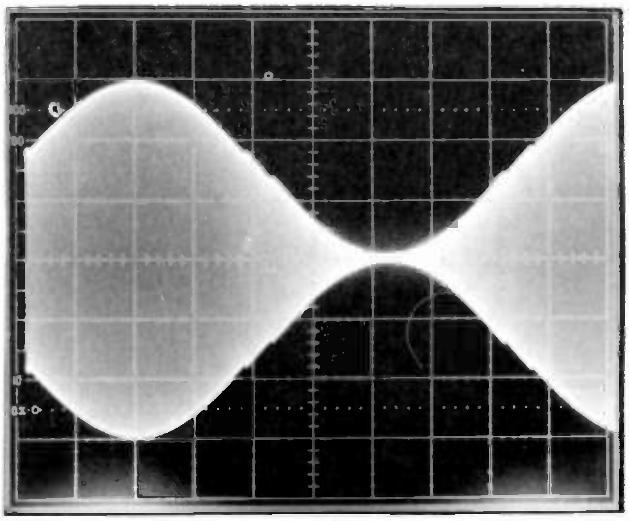
5000 Hz
Figure 8



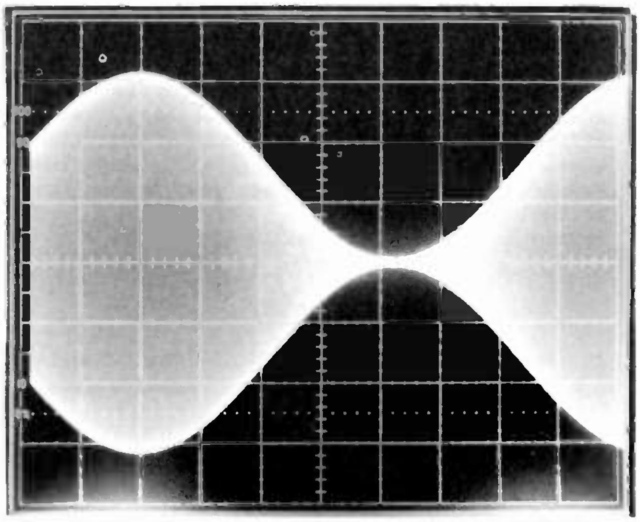
100 Hz - One Amplifier Not Operating
Figure 9



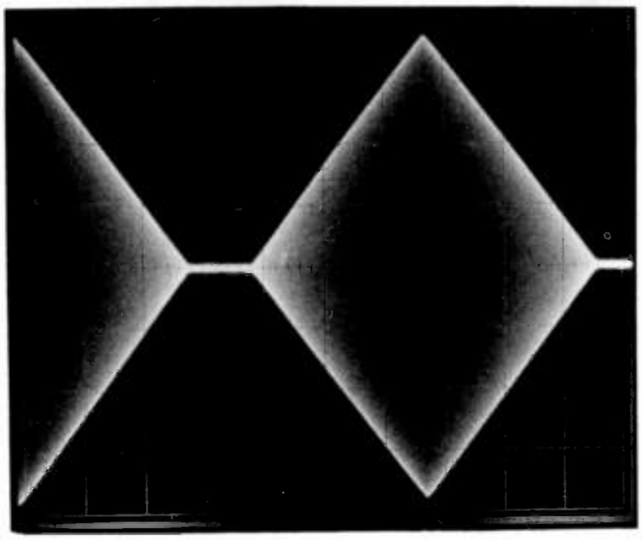
100 Hz - Binary Steps Not Operating
Figure 11



100 Hz - Two Amplifiers Not Operating
Figure 10



100 Hz - Linear Negative Peak
Figure 12



Linear Beyond 100% Modulation
Figure 13

SUMMARY

Digital Amplitude Modulation is a new and promising AM technique which is extremely efficient, reliable, and delivers excellent audio performance. Digital Amplitude Modulation is an exciting new transmitter technology to help revitalize and revolutionize AM transmission standards.

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SHARING AM TRANSMITTER SITES BY DIPLEXING ANTENNA SYSTEMS

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Introduction

The use of a single antenna structure or an antenna system of multiple structures for more than one medium wave transmitting station is hardly a novel idea. Two reference works of great durability - and considerable antiquity - Jasik's Antenna Engineering Handbook (First Edition) and Laport's Radio Antenna Engineering describe and provide feed system diagrams for multiple use of single element radiators.[1,2]

The adoption of the vertical tower radiator and the radial ground system became widespread after Gihring & Brown's "General Considerations of Tower Antennas for Broadcast Use" appeared in 1935, and Brown, Lewis & Epstein's "Ground Systems as a Factor in Antenna Efficiency" appeared in 1937.[3,4]

In marked contrast to the "flat-top" and "inverted L" antennas that had previously been the norm, the vertical tower radiator with an insulated base has a far lower "Q" and is easy to feed with simple two and three element impedance matching networks. When necessary to reduce harmonic radiation or cross-modulation products, these networks can be designed with traps or filters. A number of "diplexed" or "multiplexed" medium wave antenna systems were in operation as early as the late 1930's, although they have never been common. Early "multiplexed" antenna designs appear to have been most commonly used when two stations were under common ownership, before the prohibition of dual ownership of stations in the same market area in the U.S. A few examples survive, but most were abandoned when one of the two stations, under separate ownership, increased power or opted for a more favorable antenna site or directional antenna pattern.

A few newly devised "multiplexed" systems were constructed in the period directly after World War II, when existing stations finally gave up outmoded systems or inadequate sites. Most of these situations were the result of the lack of satisfactory sites due to cost or land use restrictions. The U.S. 5 kw restriction on power output for Class III stations reduced siting flexibility. Outside the United States and Canada, in areas where directional antennas are uncommon or even virtually unknown, fairly extensive use of multiplexed single tower omni-directional antennas became more common in the 1960's and 1970's. It should be kept in mind that, although VHF frequency modulated services have become mature and even dominant in parts of Europe and North America, in much of the rest of the world medium wave is still the preferred or even the only aural broadcast service.

In some countries single ownership of multiple stations has made multiplexed antennas a natural choice. In a few cities land use restrictions and enormous property costs have restricted the number of potential sites, causing nearly all the transmitters in the market to be consolidated into a small number of antenna systems. Honolulu is probably the best known example of this type of situation in the U.S. Other recent systems, such as that employed for the British Independent Broadcasting Authority's two London transmitters, are the result of changing regulatory circumstances, leading to the creation of new medium wave broadcasting services.[5]

In the United States in the last 10 years, a variety of regulatory and economic circumstances have led to expansion of MW services, while at the same time sharply restricting the supply of suitable antenna/transmitter sites. New nightime operation for previously daytime-only stations, and pressures to vacate sites which are valuable for other development, have led to requirements for diplexed systems.

Feasibility Analysis

A few analysis rules can be followed in considering a possible second use for an existing antenna system. Any situation where the physical characteristics of an existing antenna system would provide an acceptable solution for the second user's purposes is a possible candidate for diplexing, so long as the frequency separation between the two is about 20% or more of the lower frequency. If both facilities are omni-directional, then separation of as low as about 10 - 14% may be considered. For circumstances where maximum modulation frequencies, and consequently bandwidth, are quite low, successful systems with frequency separation as low as 3% have been constructed.[6]

If the proposed uses involve directional antennas for one or both stations then suitable patterns must be designed. Conservative criteria should be used for the directional array designs. Drive point resistances and individual element powers should not be abnormally low. Good design objectives are drive point resistances of at least 8 to 10 ohms, with at least 5% of the total power in each element. RSS values should be low, no greater than twice the RMS, and less than 1.5 X RMS if possible.

If the second user of a given site has a lower frequency than the existing one, then all of the usual factors regarding suitability of the ground system must be considered, and the FCC array efficiency requirements must be met. For an existing system where actual base impedance and mutual impedance values as well as operating impedance values for the existing user can be measured, then these should be obtained and incorporated into the feasibility analysis. The directional patterns for two frequencies need not be similar at all. Systems which include one (or more) omni-directional operation in conjunction with a directional array are also in operation. Additional towers can be added to a site to accomodate a new operation, and isolated from the existing patterns. Directional systems need not employ all of the array elements on the site, so long as unused elements are arranged to reradiate minimum amounts in accordance with the same requirements as pertain to single frequency systems.

Systems also have been constructed which are sectionalized so that there is substantially different current distribution at each frequency, in order to meet specific high angle skywave protection requirements.

Feed Systems

Antenna feed systems, often called "phasors" in the U.S., have been exhaustively studied and discussed in the technical literature. All of the elements of good design that normally apply should be considered in "diplexed" systems. In systems where a second user is added to an existing system, the compromise design of the antenna array may lead to impedance and array bandwidth limitations that are less than ideal for the second station. The feed system design itself need not be compromised. The feed system layout is generally the same as for the simple single frequency case. The diplexing circuitry is simply the addition of filters at each antenna element input, on the output leg of each antenna tuning unit network. Generally any type of feed system layout that would be advantageous for the single frequency antenna case can be used. The two feed systems need not be identical or even similar, and the branching and power dividing networks may be located where best bandwidth and efficiency are obtained.

In cases where two frequency omni-directional operation is undertaken, the feed and filtering systems reduce to the simplest situation. In situations where each station does not use all of the antenna elements, the unused ones are "floated" or detuned, as appropriate, by the filters in the feeds of the station for which a given tower is active. In diplexed directional arrays there is no requirement that the pattern or operation of one station be distorted by the electrical characteristics of the other. The only constraint is that resulting from the physical characteristics of the the antenna elements themselves - their electrical height, spacing, orientation, and number.

Filtering Requirements

There are two requirements for filtering in a diplexed antenna system. First is the isolation of each feed from the other, and second is the elimination of cross-modulation products. The first condition is imposed by the practical requirement that independent non-interactive control of the patterns at each frequency be possible. The design criterion requires that normal adjustment of each system not produce observable effects on the other. This requires (given the resolution of the usual broadband type approved monitor) 30 db isolation between systems. The approximate isolation is most easily determined by the calculation (or measurement) of the ratio of the impedance of the series filter to the impedance of the shunt arm of the associated matching network at the frequency in question.

There is no practical reason why this degree of filtering is required other than ease of independent adjustment. The filtering in omni-directional cases can be as little as 15 db without adverse effect on operation, so long as cross modulation conditions are met. The characteristics of individual transmitters vary widely, making it impossible to predict cross modulation susceptibility. Wide frequency separation generally requires less isolation than close separation. Since the electrical characteristics of antenna inputs will be different at each frequency, the base voltage ratio may be useful in calculation of the necessary filtering. The base voltage ratio in typical cases will rarely be more than 10 db, and is more typically 4-5 db.

Circuitry

A simplified example of the type of circuit that is typically used to filter an unwanted signal from the feed system of each station is shown in Figure 1.

The circuit shown in figure 1 will block a lower frequency and pass a higher frequency. The series coil and capacitor are resonant at the pass frequency and provide a capacitive reactance at the lower frequency that will be parallel resonant with the coil connected across the series circuit. An alternative scheme for achieving the same result is shown in figure 2.

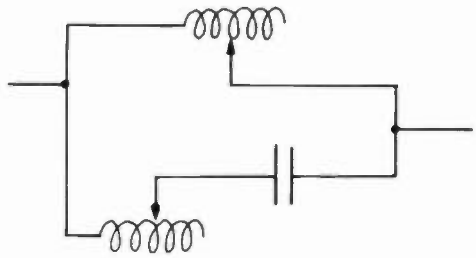


FIGURE 1

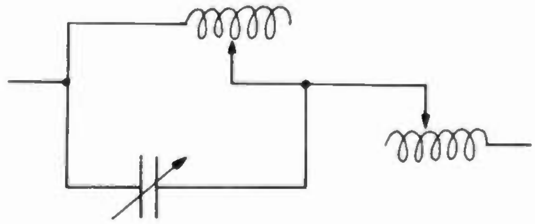


FIGURE 2

The circuit of Figure 2 passes the higher frequency by using an inductive reactance that series resonates the capacitive reactance presented by the parallel coil and capacitor at the higher frequency. The lower frequency is blocked by the resonance of the parallel components at the lower frequency.

The attenuation of the parallel resonant blocking filter is calculated from the ratio of the filter input voltage to the filter output voltage. This is comparable to the insertion loss of the filter which is calculated from the ratio of the unwanted voltage at a transmitter output with and without the filter in place. The attenuation is proportional to the circuit "Q" and inversely proportional to the magnitude of the load impedance seen by the signal that is being blocked after it passes through the parallel resonant blocking filter.

Capacitors used in filters for AM broadcast purposes have losses that are very small compared to the losses in the coils used in such filters. For this reason only the resistance associated with the "Q" of filter inductors is generally considered for the calculation of the attenuation of these networks. The parallel resonant, or tank, circuit has a very high resistance compared to its reactance at the resonant frequency. Because the resistive part of the parallel impedance at the resonant frequency is proportionately so high, the reactive component of this impedance does not have much affect on the attenuation of the tank circuit.

Calculations for a specific design illustrate the effectiveness of typical filter circuits. If coils with a "Q" (inductive reactance divided by coil resistance) of 400 are used to design a filter that blocks 1220 kHz and passes 1430 kHz, and the circuit configuration of Figure 1 is used, an L(a) of 15 uHy would require the use of an L(b) of 40 uHy and a C of 300 pF. The circuitry and relevant formulae are shown in Figure 3.

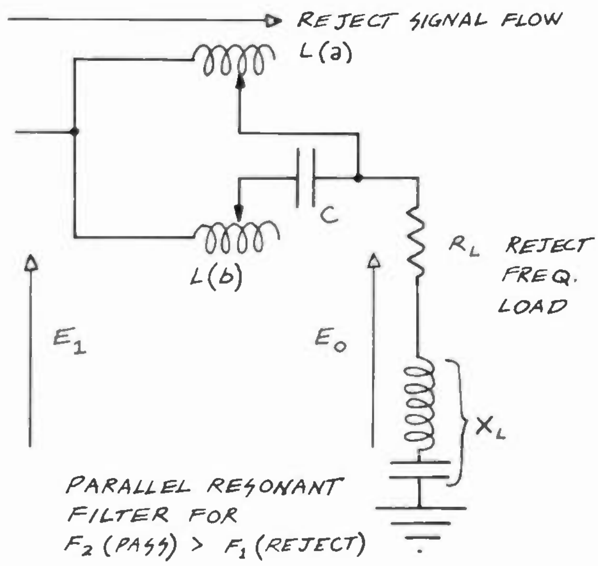


FIGURE 3

$$\text{Attenuation} = 20 \log E_1/E_o \tag{1}$$

$$E_1/E_o = (Z_t + Z_L)/Z_L \quad \text{where } Z_L = R_L + jX_L \tag{2}$$

$$Z_t = \cfrac{1}{\cfrac{1}{\frac{w_1 L(a)}{Q} + jw_1 L(a)} + \cfrac{1}{\frac{w_1 L(b)}{Q} + jw_1 L(b) - \frac{j}{w_1 C}}} \tag{3}$$

where $w_1 = 2(\text{pi})F$; F = reject frequency in mHz

and similarly for w_2 for the pass frequency

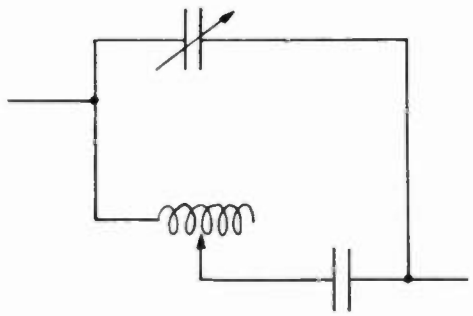
$$L(a) \times C = \left[\frac{1}{(w_1)^2} - \frac{1}{(w_2)^2} \right] \tag{4}$$

$$L(b) = 1 / [(w_2)^2 C] \tag{5}$$

where C is in microfarads, L in microhenrys

The attenuation of the Figure 3 configuration at 1220 kHz using components with these values is theoretically 48 dB if the impedance seen by the 1220 kHz signal after it passes through the filter is 50 Ohms resistive. If 50 Ohms reactance is added to this impedance the attenuation is 45 dB and decreases by six dB each time the magnitude of the load impedance at

1220 kHz doubles. If the configuration of Figure 2 is used in the design of the filter and the same 15 uHy value were used for L(a), the value for L(b) would be 40 uHy while C would be 0.0011 uF. The comparable attenuation at a 1220 kHz load impedance of 50 Ohms resistance is 59 dB. This filter has a narrow pass band and might



PARALLEL RESONANT FILTER FOR F_1 (PASS) $<$ F_2 (REJECT)

FIGURE 4

be considered in a design with close frequency spacing. The attenuation of this circuit is 50 dB when the 1220 kHz load is 100 Ohms resistance and 100 Ohms reactance and also decreases six dB with each doubling of the 1220 kHz load impedance. This demonstrates the degrading influence of increasing load impedance at the unwanted frequency upon the performance of the parallel resonant filter. Where a large attenuation of the unwanted signal is required a series resonant filter to ground across the transmitter side of the parallel resonant filter may be required.

Figure 4 shows a filter of the type of Figure 1 and 3 for use when the pass frequency is lower than the reject frequency.

Figure 5 shows a typical ATU and filter layout using the same type of filter design as Figure 1 for a situation where the frequency separation is large. Note the series capacitor at the filter output, used to compensate for the large antenna reactance so that voltages on the filter and matching networks are not abnormally high.

Figure 6 shows a system designed for a low impedance antenna with very close frequency spacing - less than 10% of the lower frequency.

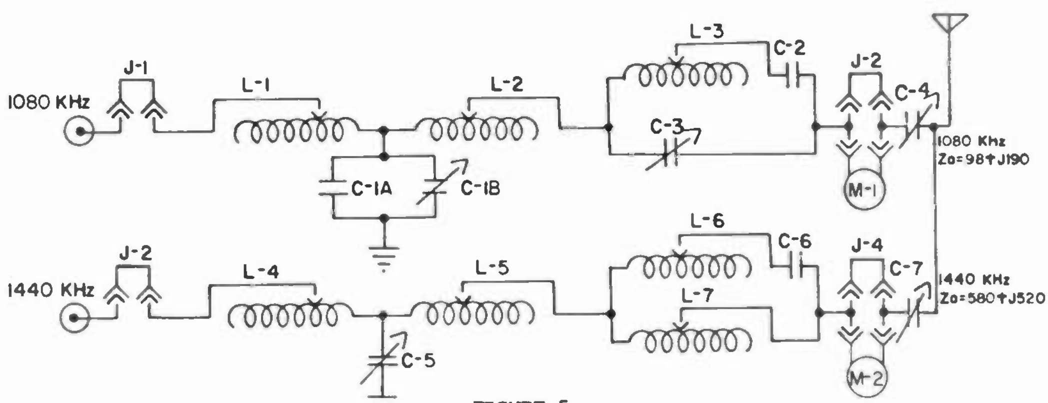


FIGURE 5

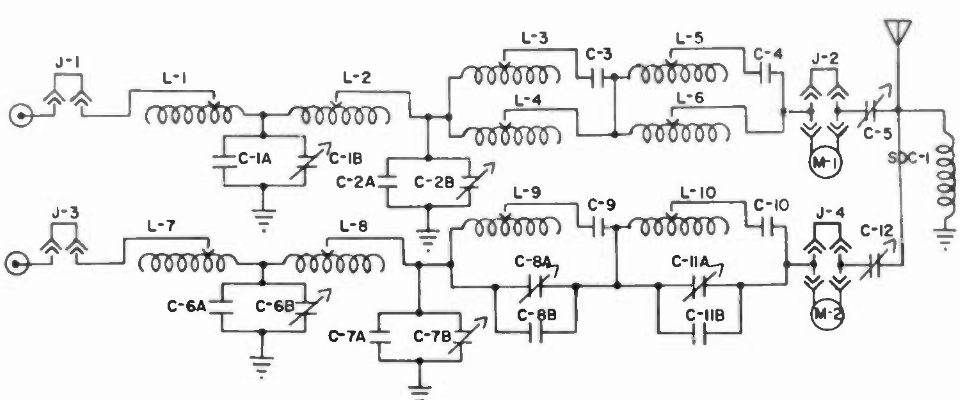


FIGURE 6

Note that two-section filters are necessary, and that a two step network is used for impedance transformation, but that even in this extreme case no shunt filters are used.

Bandwidth

The effect of the insertion of the appropriate filtering on the common point impedance is dependent on a large number of variables. The relationship of drive point voltages at each tower, which determines the filtering necessary, is the primary variable; it cannot be changed without some change in the geometry or electrical characteristics of the arrays. However, even substantial supression requirements need not produce undesirable effects over the passband. As an example the following table shows the common input impedance of a three tower directional array with a well designed phasor, together with data for the same array after diplexing a second station into the three antenna elements.

	Freq.	Z	VSWR
Before	Carrier	50 ± j0	1.00
	+ 10 kHz	50 ± j0	1.00
	- 10 kHz	49 ± j3	1.056
After	Carrier	50 ± j0	1.00
	+ 10 kHz	50 ± j5	1.040
	- 10 kHz	49.5 - j1	1.023

Cross Modulation

In most instances the filtering necessary to allow independent monitoring and adjustment of the antenna parameters for each antenna pattern will also be adequate for protection against the generation of cross modulation components in the transmitters. When this is not the case, or when other nearby sources of radio frequency energy are present to the degree necessary to require filtering, it has been found most desirable to incorporate the filtering in the shunt leg of the common input matching network. In some instances it would also be possible to incorporate such filtering in the shunt elements of matching networks in the transmitter itself. In each instance the choice should be made on the basis of its effect on the impedance bandwidth as seen by the output stage of the transmitter insofar as it is consistent with the requisite degree of cross modulation suppression. Some modern transmitters with wideband output stages may require more filtering than older types. Transmitters with extensive output filtering such as most PDM types appear to be quite immune to such difficulties.

Monitoring Systems

It has been our practice to use monitoring systems which obtain samples in the output arm of each filter element. In cases where the antenna elements are over 120 degrees in length at one or both frequencies, proper samples at the current maximum (as defined by traditional current distribution analysis) may be desirable. The loops now generally used are low inductance one turn loops which are about equally sensitive to both sets of current in the tower. Therefore the sample must be filtered after it is brought across the tower base (with the sample line isolation coils either adequately inductive at both frequencies or detuned at both). When filters are in use in the sample system, they must be arranged so that they can be bypassed during maintenance periods, or secondary samples must be taken in the output arms of the filters. Use of filters in the sample system may allow a corresponding reduction in the feed system filtering requirement, if cross-modulation conditions are met. The feed system filters should, however, be designed so that no change in current at one frequency can be observed when normal adjustment of the feed system for the other frequency takes place.

Setup and Adjustment

a. The first step in adjustment of any directional feed system or omni-directional matching network is to measure the self impedance of each element and the mutual impedances at the point of junction of the filter networks. Then all network values should be recalculated and all branching (power division) and matching networks other than the antenna match networks set to the calculated values.

b. The output arms of the antenna match networks will include the stray impedances of the filters. Therefore the filters should be set before the tower match networks. For the usual configuration, the filter zero net (series resonant pass networks) are set first, the parallel resonant reject portions are then adjusted by connecting them as series nets and adjusting for zero. When reconnected as parallel "tanks", adjustment can be trimmed using a signal generator and field set in conjunction with a toroidal transformer. Once each filter is preliminarily adjusted, the entire antenna match network associated with it can also be adjusted.

c. At this point the systems can be adjusted for the correct (calculated) phase and current ratios. Preliminary monitor point or other field observations may be made.

d. When the system has been trimmed to preliminary phase and current ratio values, the individual filters can be adjusted for exact parallel resonant conditions. Using current transformers (or a sample taken with a divider at the match network junction) the filter can be adjusted for best rejection. The series resonant portion of each filter should not require any adjustment at this point in the process.

e. If the system has been constructed with toroidal samples on each filter output, the sample system will allow determination of proper system adjustment - each sample system should show no variation as the other transmission system is turned on and off, or as small adjustments are made to its phase and current ratios.

f. Monitoring proper performance in a transmission system with filtering in the sample system is not nearly as simple. Without some special provision for monitoring, the transmission system cannot easily be tested during normal operation. So proper performance is most easily checked by disconnecting the feed to each antenna match network and comparing the rejected signal value with a value logged at initial setup, or by including extra toroidal samples in the system. A sample can also be taken at the transmitter modulation monitor output but this does not allow an abnormal condition to be easily traced to a particular antenna element filter.

Conclusion

The multiple use of medium wave antenna systems is likely to increase in the future. The elements of design for these systems are straightforward and can be solved for many practical circumstances. Careful design and attention to detail during measurement and setup result in systems which are stable and practical.

Acknowledgements

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USING HELICOPTER MEASUREMENTS TO EVALUATE SOURCES OF RE-RADIATION OF AM STATIONS

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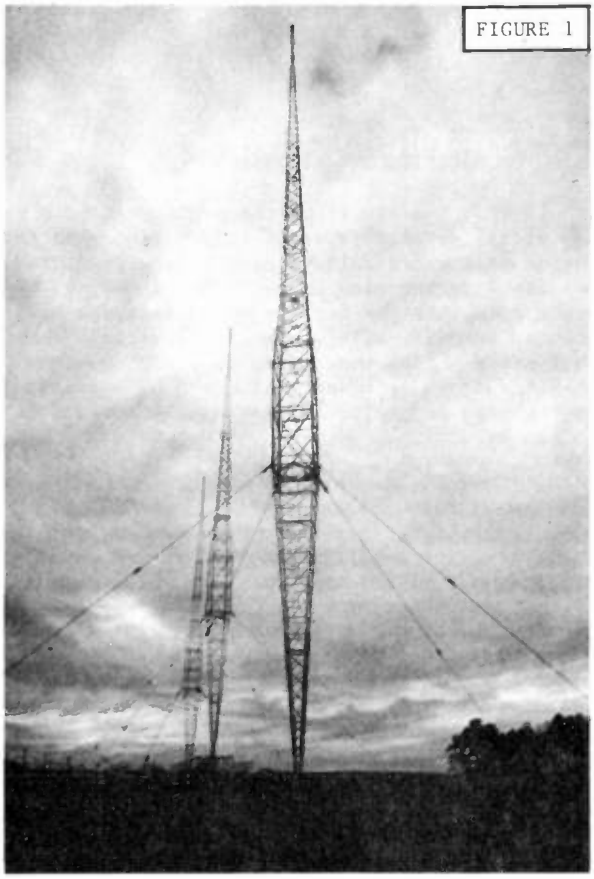
INTRODUCTION

AM broadcast stations, and particularly directional AM's, must deal with a phenomenon called "reradiation", the undesired and uncontrolled radiation of the station's signal by objects external to the antenna system and often not under the licensee's control. In this paper, the effect of reradiation is discussed briefly, and a means of quickly and efficiently investigating a reradiation situation is demonstrated.

FIGURE 1



AM towers in the vicinity of another station are always likely reradiators.

WHAT IS RERADIATION?

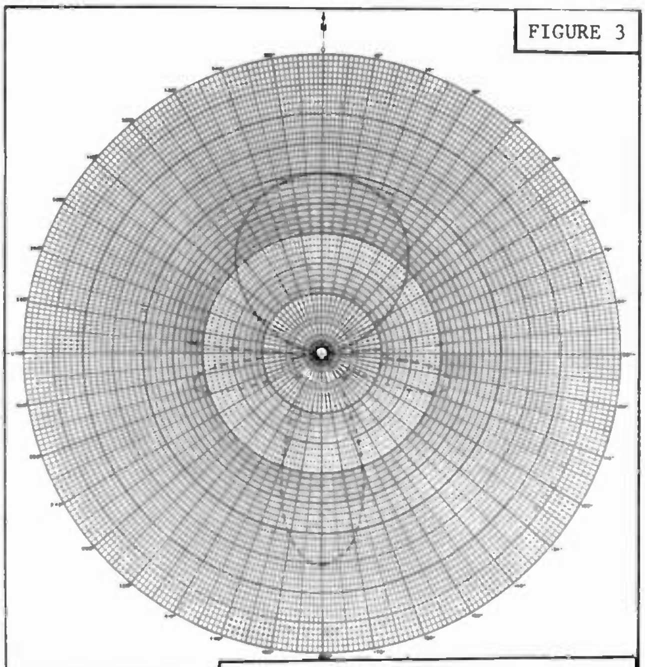
Reradiation occurs when a conducting object of a suitable size is exposed to an RF field of sufficient strength to cause RF currents to flow on the object. These RF currents cause radiation in precisely the same manner as current flowing in any antenna. Sometimes, as in the case of a YAGI array, this is useful. YAGI antennas generally employ one driven element and several parasitic "directors" and "reflectors". As used in this paper, the term "reradiation" will refer to the undesired radiation from objects which are not intended to radiate and are excited incidentally.

FIGURE 2

Construction cranes present a unique problem because they move around on a single spot and from place to place.

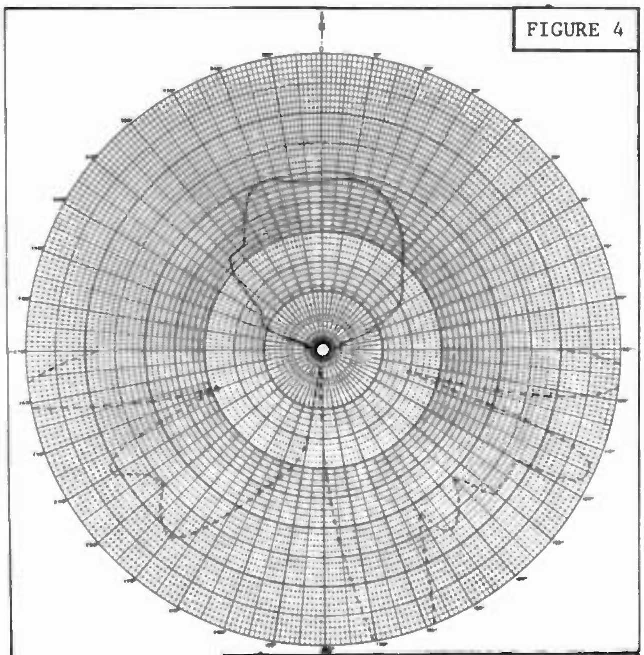
A reradiator need not look like a tower. Buildings have been shown to reradiate[1] as have cranes, stadiums, bridges, smoke stacks and other structures. Figures 1, 2, 5, 7, 14 and 15 are typical of the kinds of structures which may be expected to reradiate. Nor must it be located on the station's property. Often, the most troublesome reradiators are those located at some distance from the array on someone else's property. Under these conditions, the solution to the problem may involve more diplomacy than technology.

FIGURE 3



That a solution to the problem must be found is illustrated by Figures 3 and 4. Figure 3 is the theoretical pattern of a four tower in-line array. Also shown is a 20:1 expanded scale plot of the null structure. Note that the array offers high gain to the north and very deep suppressions to the south. Figure 4 shows the distortion which can result from the addition of a reradiating object placed at a distance of 1000 degrees (about 1/2 mile) from the array and reradiating less than 5% of the incident field.

FIGURE 4



The desired null structure has been completely destroyed and there is considerable null fill. In addition, there is considerable distortion of the major lobe which is both scalloped and somewhat re-oriented.

FIGURE 5

High tension power lines and their support structures can create complex reradiation problems.

LOCATING THE RERADIATORS

During the course of the adjustment of a new array, we might suspect reradiation when the design parameters fail to produce the predicted results. In the case of an operating array, we might find that the monitor point readings have changed suddenly with no change in array parameters. In either case, we would conclude that our array is being influenced by external factors and begin looking for likely reradiators in the area.

However, especially in urban areas, it's not always easy to identify which objects, if any, may be contributing to the problem. In fact, it is often difficult to determine whether a particular reradiation problem is the result of one reradiator or several. And, as has been mentioned, reradiators don't always take the expected shape. In fact, virtually any thing which conducts and which is of significant size with respect to a wavelength may become a reradiator.

The treatment of reradiators is, at best, a time consuming and expensive proposition. Where the reradiator is not under the control of the licensee, negotiations to allow the detuning usually involve considerable time as well as expense, with no guarantee of complete success. Therefore, we must be certain that the structures to be detuned actually require detuning, that proper detuning methods are employed, and that the detuning has been effective.

A number of procedures have been used to attempt to locate and quantify reradiation problems. One commonly used procedure involves taking measurements along a segment of a circle which has as its diameter a line connecting the source and the suspected reradiator. (See Figure 6.) From any point on that circle, a line to the source and a line to the reradiator are at right angles to each other. The field intensity meter's loop antenna has a deep null broadside to its axis, so at each point on the circle, if the antenna is oriented toward the reradiator, its sensitivity is minimized toward the source and vice versa.

FIGURE 6

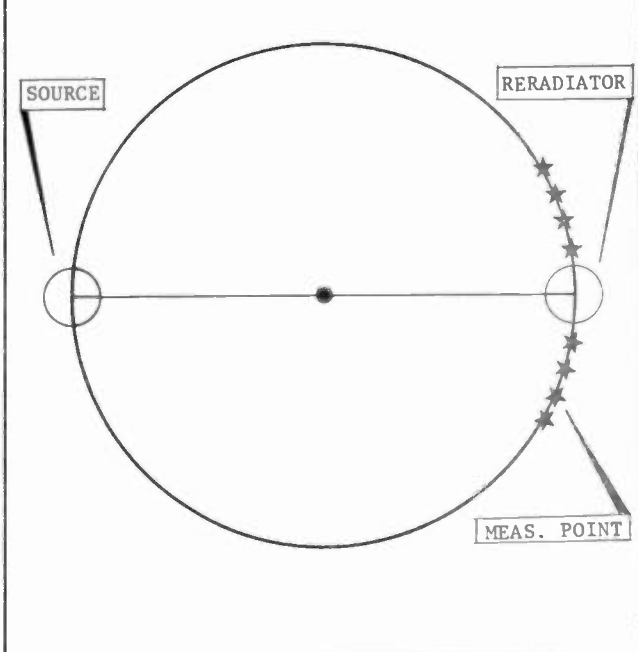


FIGURE 7

Many AM engineers have been surprised by the sudden appearance of a Cellular radio tower nearby.

Another approach to reradiation measurements involves taking readings at frequent intervals along a radial between the array and a suspected reradiator. As we will see, the presence of reradiation is indicated by a pronounced standing wave pattern the amplitude of which is a measure of the magnitude of the reradiation.

THE STANDING WAVE

The classic indication of reradiation is the presence of a standing wave pattern in measurements taken along a line connecting the source and the reradiating object. This is illustrated in Figure 8, a graph of the fields which might be expected with a source radiating 100 mV/m/mi and a reradiator at a distance of one mile which radiates 10% of the incident field or 10 mV/m/mi. The computer program which generated these data was written to assume measurements taken at an altitude of about 500' and makes the appropriate correction for the vertical angles. A frequency of 931.411 KHz. has been selected to avoid odd intervals. At that frequency, one mile equals 5 wavelengths or 1800 degrees and one wavelength equals 0.2 miles. As can be seen, the standing wave pattern begins abruptly as the observer, in all cases traveling from right to left, passes the reradiator and is between the source and the reradiator. In addition, although this may not be immediately obvious, the period of the standing wave pattern is equivalent to one half that of the RF signal we're measuring. The reason for this will soon become clear.

Let's look in detail at the fields at 7 points along the observer's path, again, moving from right to left. Figure 9 is, in part, an expansion of Figure 8 showing a portion (0.8 to 1.1 miles) of the distance between the source and the reradiator. Point A is at a distance of 1.1 miles from the source and 0.1 miles from the reradiator. Point B is at 1.05 miles from the source. These points, at our chosen frequency, are 90 degrees apart. Thus, as the observer proceeds from Point A to Point B, the distance to both the source and the reradiator decreases by 90 degrees.

Figure 9A1 is a representation of the fields which will be present at point A. They are shown as vectors, the first being the field from the source and the second, the field from the reradiator. Figure 9B1 shows the same conditions with respect to point B. Point A is at a distance of 1.1 miles or 1980 degrees, some 5.5 wavelengths from the source. At point B, the distance is 1890 degrees. As the observer draws nearer the source, the phase delay due to distance from the source decreases, and that vector rotates in a counterclockwise direction as shown. Since the distance to the reradiator

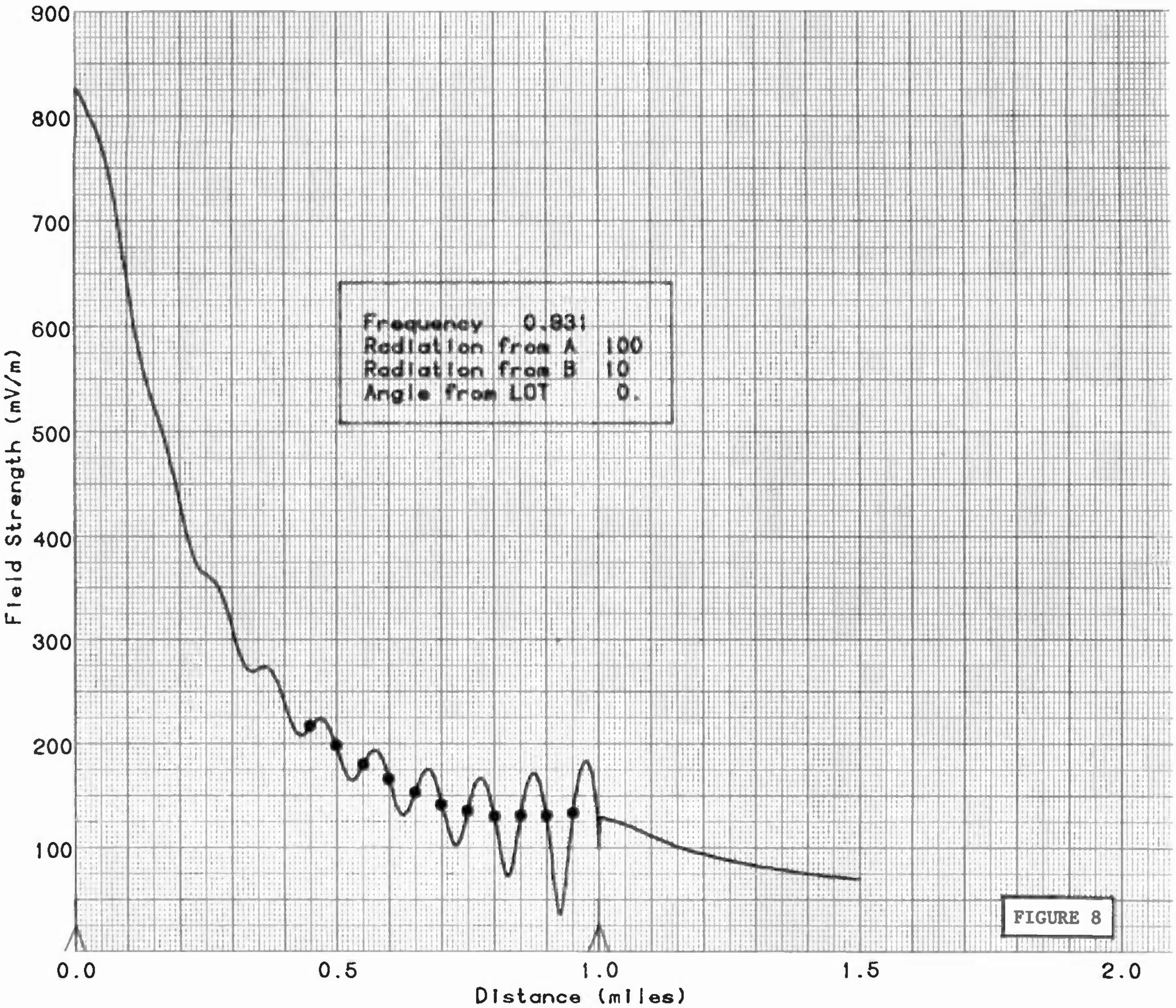


FIGURE 8

decreases at the same rate, both vectors rotate the same amount in the same direction and the phase relationship between them is unchanged. This is shown in Figures 9A2 and 9B2 in which the source vector is chosen to be the phase reference.

As the observer passes the reradiator and continues toward the source, the vector representing the source field continues to rotate counterclockwise as before. However, now as he approaches the source, he increases his distance from the reradiator. The effect of this is to cause the vector representing the reradiated field to rotate clockwise, reversing its direction of rotation. Again making the field from the source the phase reference (Figures 9C2 to 9G2), the reradiated field vector can be seen to rotate about the end of the source field vector. It is this rotating vector which creates the tell-tale standing wave pattern. It is important to note that for every degree nearer the source, the observer is also one degree further from the reradiator. Thus, for each degree the observer travels, the vectors will each rotate one degree but in opposite directions creating a 2 degree relative phase change. Therefore, the period of the standing wave is 1/2 the wavelength at the operating frequency or 0.1 miles.

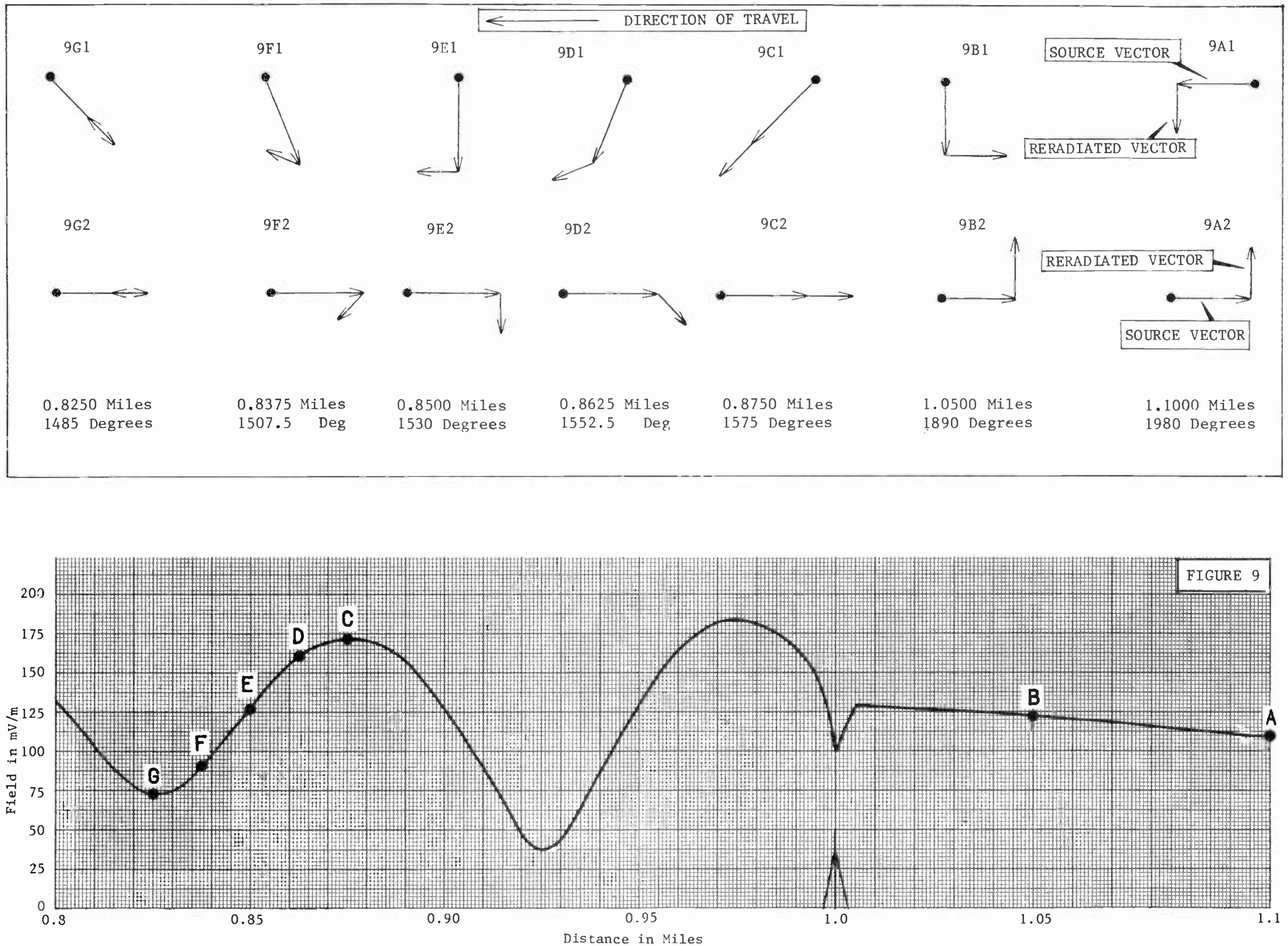

DIRECTION OF TRAVEL
9G1
9F1
9E1
9D1
9C1
9B1
SOURCE VECTOR
9A1
RERADIATED VECTOR
9G2
9F2
9E2
9D2
9C2
9B2
RERADIATED VECTOR
9A2
SOURCE VECTOR
0.8250 Miles
1485 Degrees
0.8375 Miles
1507.5 Deg
0.8500 Miles
1530 Degrees
0.8625 Miles
1552.5 Deg
0.8750 Miles
1575 Degrees
1.0500 Miles
1890 Degrees
1.1000 Miles
1980 Degrees
Field in mV/m
200
175
150
125
100
75
50
25
0
G
F
E
D
C
B
A
0.8
0.85
0.90
0.95
1.0
1.05
1.1
Distance in Miles

FIGURE 9

RERADIATION MEASUREMENTS

We have previously mentioned two common measurement procedures in reradiation situations. Both techniques are time consuming and prone to error. For example, circle segment measurements depend on the ability of the loop antenna to achieve a complete null broadside to the loop's axis. In areas of high incident field, the relatively lower fields from the reradiator may well be masked by the incident field. This is particularly true if extreme care isn't taken in orienting the meter. And, if either the source or the reradiator is not a simple structure (an AM directional array, for example), it probably will not be possible to achieve the required orientation.

Making extensive ground measurements along a radial which includes both the source and the suspected reradiator will show the standing wave pattern described above and is the other common method of evaluating reradiation. However, this technique requires that a considerable number of measurement points be taken. Looking at Figure 8, we can see that if measurements were taken at intervals of 0.05 miles or 264' (represented by the dots), it is virtually certain that the standing wave pattern will be overlooked. Even with a measurement interval of 100' or so, it is highly unlikely that the points would include both the maximum and minimum excursions of the field. The only way around this is continuous metering of the field, as with a chart recorder.

The problem is further complicated by the likelihood that the nature of the terrain in the area of the transmitter site will make measurements at sufficiently short intervals difficult if not impossible. In developed areas, we will probably find homes or commercial buildings in the path while in rural areas, the path may be heavily wooded or swampy or otherwise impassable.

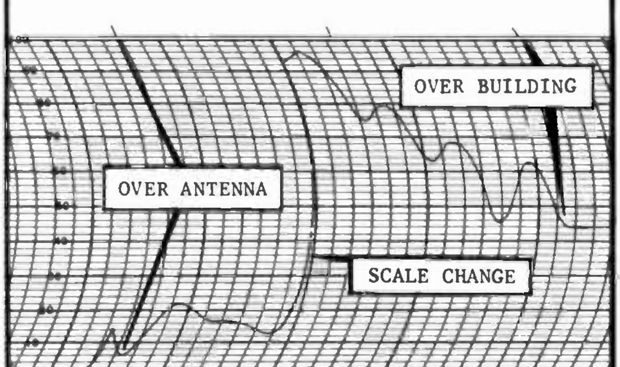


FIGURE 10

The use of a helicopter to traverse the radial, and a chart recorder to log the field can effectively overcome these problems. Making the measurements by helicopter offers the additional advantages of speed and an unobstructed view of the area around the site so that potential reradiators can be more easily identified. Figure 10 is a reproduction of the chart of an actual helicopter run over a reradiating building. The marks at the top of the chart were made with the chart recorder's "event" pen and correspond to terrain features and landmarks such as creeks, railroad tracks and highways. By use of these "ticks" and appropriate interpolation, it is possible to determine the distance from the reradiator at each point along the chart. Like the previous figures, the direction of travel is right to left. A pronounced standing wave pattern begins abruptly as the reradiating building is overflown and damps out as the source is approached.

In addition to establishing whether or not a particular structure reradiates, the magnitude of the reradiation can be found from these measurements. Looking first at Figures 9C2 and 9G2, it is clear that the difference between the maximum and minimum field is equal to twice the amplitude of the reradiated vector. Conversely, the magnitude of the rotating vector is one half of the difference between the maximum and minimum fields.

Figure 11 is the same graph as Figure 8 but with dotted lines drawn to approximate the envelope of the standing wave. Close to the reradiator the vertical angle is significant due to the altitude of the helicopter. But, if we look, for example, at 0.3 miles from the reradiator, we find that the envelope extends from 107 to 173 mV/m. The difference is 66 mV/m so the length of the vector is 33.0 mV/m. Assuming inverse distance propagation, the radiated field can be found from:

$$E_0 = E*D$$
$$= 33.0*.3$$
$$= 9.9 \text{ mV/m/mi.}$$

At distances of:

	0.4 miles	0.5 miles
Maximum	= 187	215
Minimum	= 137	175
envelope	= 50	40
env/2	= 25	20
E*D	= 25*.4 = 10	20*.5 = 10

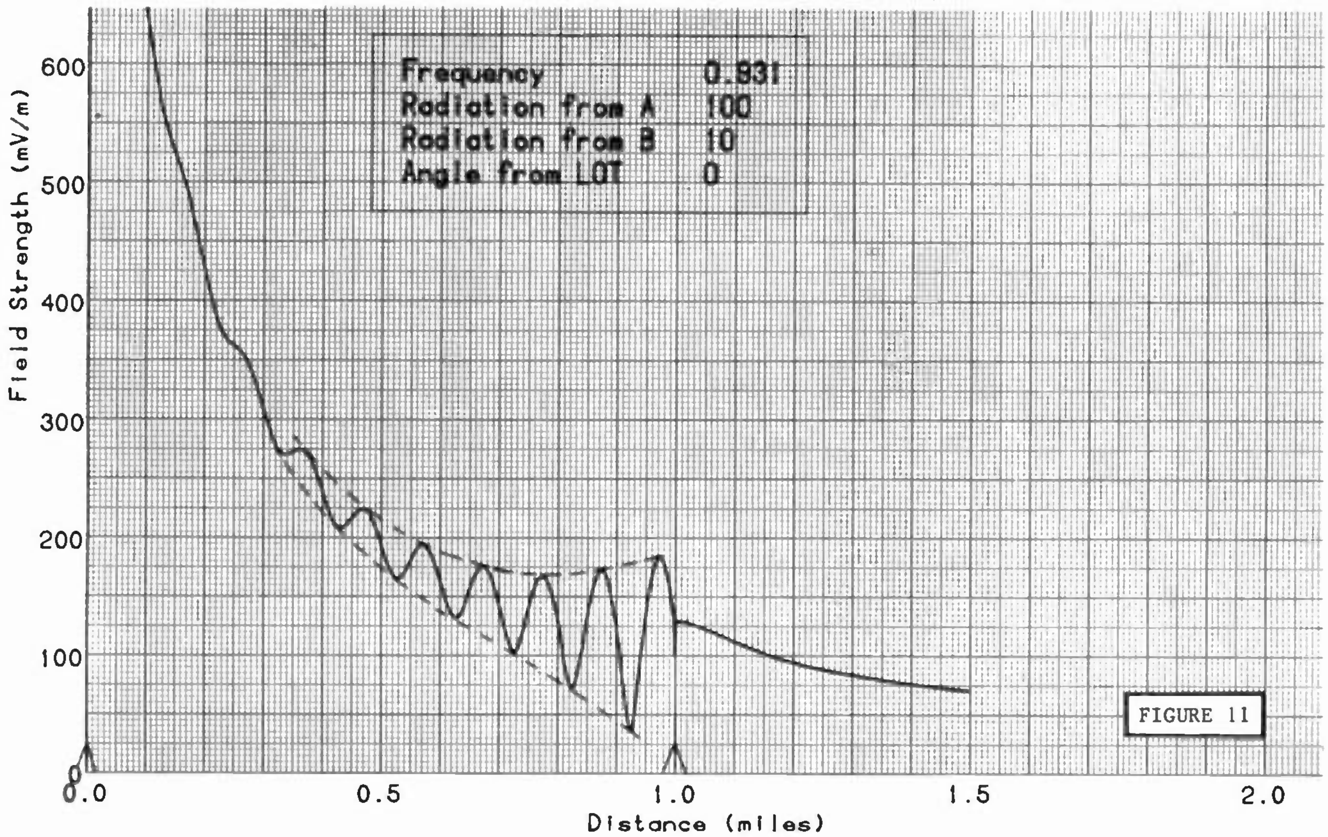


FIGURE 11

The original value of reradiated field was established at 10 mV/m/mi. Clearly this technique permits the magnitude of the reradiation to be established with considerable accuracy. In the case of field data, we generally make these calculations at several distances and average the result to minimize the effect of errors in reading the chart and other possible errors.

OBJECTS OFF THE RADIAL

So far, the discussion has been limited to the results to be expected when the flight path is a radial which includes both the source and the reradiator. It is not uncommon, however, to encounter reradiators which may be near but not on the radial. Figure 12 is a graph showing a situation in which the reradiator is located at a distance of one mile from the source but is 30 degrees off the measured radial. (The assumed reradiation has been increased to 15 mV/m to make the effect more visible.) As can be seen, the standing wave pattern, rather than starting abruptly at the reradiator, begins before the object is passed. In addition, the period of the pattern, rather than being constant, is long near the reradiator and grows shorter as the source is approached. This is caused by the fact that the rate of change of distance from the reradiator changes along the path of the radial. A pattern like this is a clear indication of a reradiator near but not on the flight path. Figure 13 is a tracing of an actual measured radial over a radio tower which was suspected of reradiation. As can be seen, no significant change in the readings took place as the tower was overflown, suggesting that it was effectively detuned. However, at the point marked "A", a standing wave pattern similar to that of Figure 12 had already begun. And yet another standing wave pattern starts near point "B" closer in. This run led to two conclusions, first that the radio tower was not a problem and, second, that there were 2 structures in the area which warranted further study. Figure 10 is, in fact, a subsequent run over the closer of those two structures, a building which is shown in the photograph in Figure 15. Figure 14 is the building at point "A" on the graph.

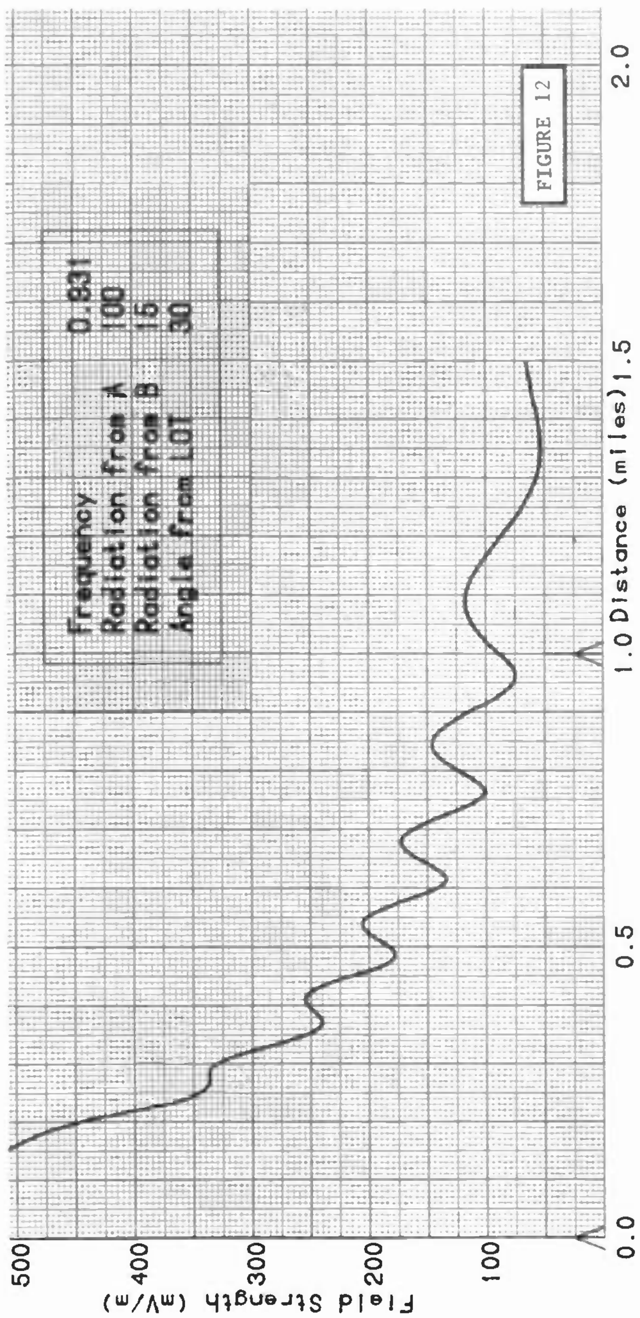


FIGURE 12

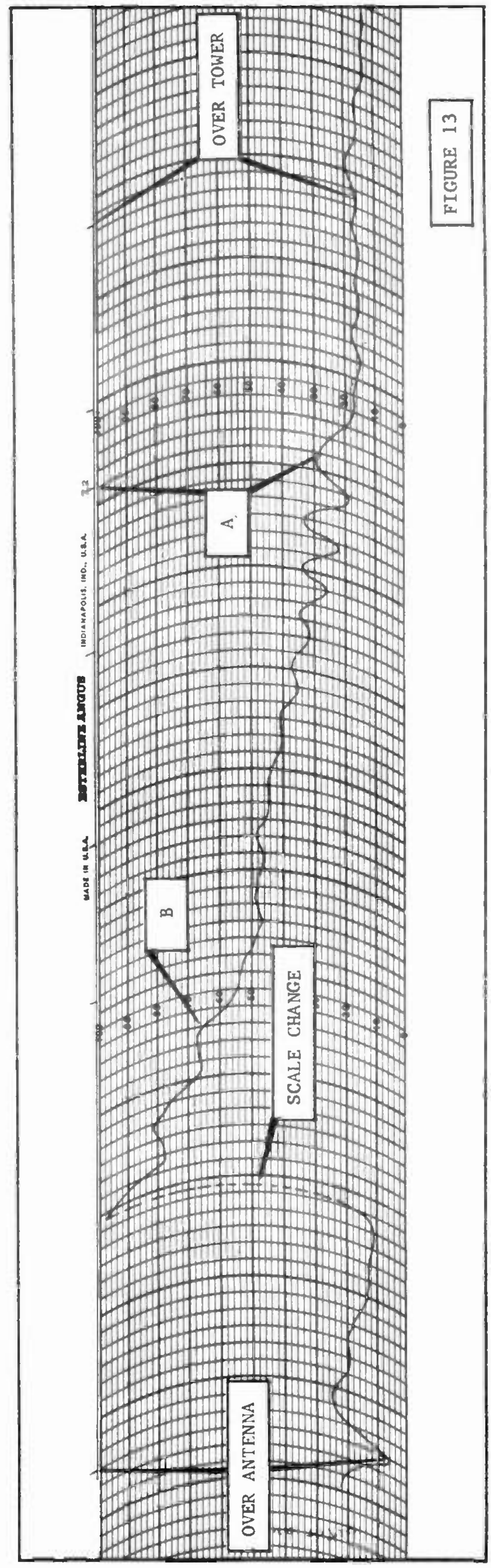


FIGURE 13

FIGURE 14

Buildings, such as shown above and below, have been shown to be significant reradiators. Detuning of these buildings is difficult due to their size and the fact that the steel framework is imbedded in a lossy dielectric.

FIGURE 15

FIGURE 16

A Hughes 300C helicopter ready for a reradiation run. The chart recorder and field meter are visible.

EQUIPMENT SET-UP

Figure 16 is a photograph showing the equipment in place in the helicopter ready to begin the measurements. The helicopter is a Hughes 300C, which was selected because the rental cost is less than half the cost of a jet helicopter and because it is small and maneuverable. The field intensity meter is a Potomac Instruments FIM-41 and an Esterline-Angus chart recorder was used. The chart recorder uses a wind-up clock motor and the field meter operates on batteries so no connections to the helicopter's electrical system are required. Since all the equipment is inside the aircraft, no airframe modifications are required. The field meter's antenna was inside the aircraft so a correction factor was used to adjust the measured fields for the effect of the aircraft. The calibration run for this purpose involved taking readings on the ground with the helicopter out of the immediate area and comparing those readings with those taken in the aircraft hovering over the same points. A number of points were taken, and the correction factor was the average of the ratios of the ground measurements to those taken in the helicopter.

OTHER USES

With good air to ground communications, this method can also be employed to adjustment detuning apparatus. Following each adjustment of the detuning, a radial, or a portion thereof, is flown over the structure. The process is continued until the amplitude of the standing wave pattern is reduced to a minimum.

SUMMARY

It has been shown that helicopter measurements can be useful both in identifying reradiators (as well as those objects which do not reradiate) and in quantifying the magnitude of the reradiation. In addition, objects near the flight path which reradiate can also be identified and evaluated. Because of the speed of the helicopter, it is possible to investigate a number of structures in a relatively short time. Since the logging by chart recorder is continuous, where there is reradiation, the pattern is clear and unmistakable and can be readily seen during the course of the measurement run. The detailed analysis of the charts can be performed later on the ground, thereby minimizing the necessary helicopter time.

ACKNOWLEDGMENTS

I would like to thank my employer, Moffet, Larson and Johnson, Inc. both for the time to write this article and for the considerable technical and editorial help I received. The particular contributions of Frank Aghili to the graphics and layout are especially appreciated as are those of my good friend and teacher, Ogden Prestholdt.

[1] A great deal of work in this area has been done in Canada, especially at the University of Toronto and Concordia University in Montreal. With regard to building reradiation, the article "Highrise Building Reradiation and Detuning at MF" by Kavanagh and Balmain which appears in the 1 March, 1984 issue of IEEE Transactions On Broadcasting, Vol BC-30 is of particular interest.

PROTECTING A BROADCAST FACILITY FROM TRANSIENT DISTURBANCES

Jerry Whitaker
Broadcast Engineering Magazine
Overland Park, Kansas

Every electronic installation requires a steady supply of clean power in order to function properly. The ac power line into broadcast plant is -- in fact -- the lifeblood of any operation. It is also, however, a frequent source of equipment malfunctions and component failures. Recent advances in technology have made the question of ac power quality even more important, as microcomputers are integrated into a wide variety of broadcast products. The high speed-logic systems prevalent today can garble or lose data because of power supply disturbances or interruptions.

<u>The scope of the problem</u>

The utility company feed into broadcast plant contains not only the 60Hz power needed to run the facility, but also a variety of voltage abnormalities. These disturbances cause different types of problems for different types of equipment.

Figure 1 shows the four major classifications of short-term ac voltage disturbances. The generally-accepted definitions for these disturbances are:

1. Voltage surge -- An increase of 10% to 35% above the normal line voltage for a period of 16ms to 30 seconds.

2. Voltage sag -- A decrease of 10% to 35% below the normal line voltage for a period of 16ms to 30 seconds.

3. Transient disturbance -- A voltage pulse of high energy and short duration impressed upon the ac waveform. The overvoltage pulse may be 1 to 100 times the normal ac potential and may last up to 15ms. Rise times can measure as short as 1 nanosecond.

4. Momentary power interruption -- A decrease to zero voltage of the ac power line potential, lasting from 33ms to 133ms. (Longer duration interruptions are considered power outages.)

Voltage surges and sags may occasionally result in operational problems for equipment on-line, but generally automatic protection or correction circuits will take appropriate actions to ensure that there is no equipment damage. Such disturbances can, however, garble computer system data if the disturbance transition time (the rise or fall-time of the disturbance) is sufficiently fast. System hardware may also be stressed if the power supply has marginal reserve or the frequency of disturbances is great.

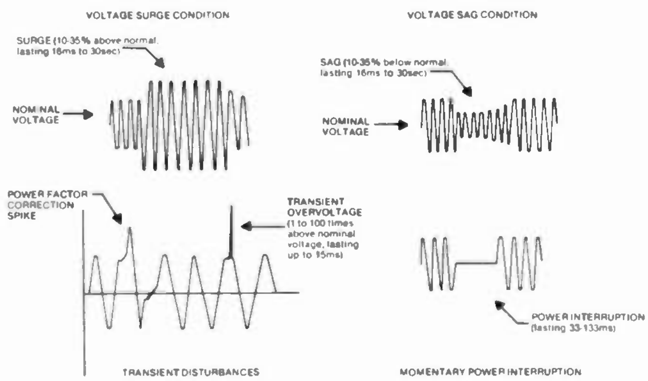


Figure 1. The four basic classifications of short-term power-line disturbances.

Momentary power interruptions can cause a loss of volatile memory in computer-driven systems and severely stress hardware components, especially if the ac supply is allowed to surge back automatically without soft-start provisions. Successful system reset may not be accomplished if the interruption is sufficiently brief.

Although voltage sags, surges and momentary interruptions can cause operational problems for equipment used today, the threat of complete system failure due to one of these mechanisms is relatively small. The greatest threat to the proper operation of broadcast equipment rests with transient overvoltage disturbances on the ac line. Transients are difficult to identify and difficult to eliminate Many devices commonly used to correct for sag and surge conditions, such as ferro-resonant transformers or motor-driven autotransformers, are of limited value in protecting a load from high-energy, fast rise-time disturbances on the ac line. If not attenuated, these brief pulses -- sometimes only a few microseconds in duration -- can destroy semiconductors, disturb logic operations or latch-up microcomputer routines.

Experience in the computer industry has shown that the vast majority of unexplained problems resulting in disallowed states of operation are actually caused by transient overvoltages on the utility feed. And with the increased use of microcomputers in broadcasting, this warning cannot be ignored. The threat to broadcast facilities is compounded by the fact that microcomputers are being used at critical stages in the transmission chain, including program automation equipment and transmitter control systems.

Because of the high potential that transient disturbances typically exhibit, they can cause not only data and program errors, but also result in damaged or destroyed electrical components. This threat involves both sensitive integrated circuits and many other common devices as well, such as capacitors, transformers, rectifiers and power semiconductors. Figure 2 illustrates the vulnerability of common components to high-energy pulses. To compound the threat, the effects of transient disturbances on electronic devices are often cumulative, resulting in gradual deterioration of the device and -- ultimately -- catastrophic failure.

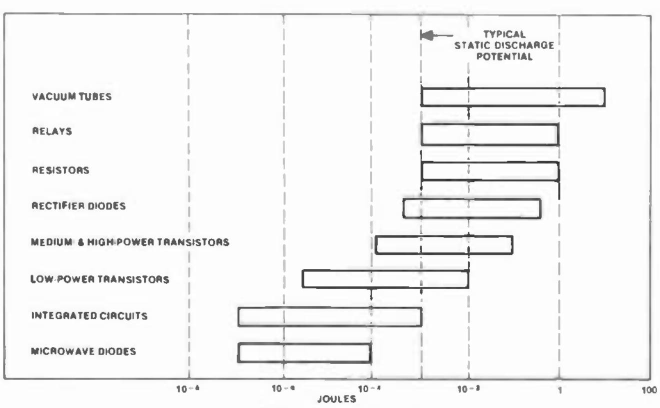


Figure 2. An estimation of the susceptibility of common electrical devices to damage from transient disturbances. The vertical line marked "static discharge" represents the energy level of a discharge that typically can be generated by touching a piece of equipment after walking across a carpeted floor. (Reference 1.)

Protection alternatives

Most utility companies make a good-faith attempt to deliver clean, well-regulated power to their customers. Most disturbances on the ac line are beyond the control of the utility company. Lightning strikes, large load changes imposed by customers on a random basis, power factor correction switching and accident-related system faults all combine to produce an environment in which tight control over ac power quality is difficult -- at best -- to maintain. The responsibility, therefore, for ensuring ac power quality must rest with the user of the sensitive equipment.

The selection of a protection method for a given facility is as much an economic question as it is a technical one. A wide range of power line conditioning and isolation equipment is available. A logical decision on how to proceed can only be made with accurate, documented data on the types of disturbances typically found on the ac power service to a given facility.
This data can be gained from a power quality survey, available from a number of consulting firms and power conditioning companies. The typical procedure involves installing a sophisticated voltage monitoring unit at the site to be protected for a period of several weeks, during which data is collected on the types of disturbances that the load equipment is likely to experience.

The type of monitoring unit used is of critical importance. It must be a high speed system that stores disturbance data in memory and delivers a printout of the data on demand. Slow-speed chart recorders, used by many utility companies, are far too slow and lack sufficient sensitivity to accurately show short-duration voltage disturbances. Chart recorders are useful for confirming the presence of long-term surge and sag conditions (of 10 seconds or more), but provide almost no useful data on transients.

The protection equipment chosen must be matched to the problems found to exist on the line. The use of inexpensive basic protectors may not yield much improvement over operating directly from the ac line. Conversely, the use of a sophisticated protector designed to shield the plant from every conceivable power disturbances may not be economically justifiable.

The initial cost of purchasing transient suppression equipment is only one element in the selection equation. Consider the costs associated with site preparation, installation and maintenance. Also consider the operating efficiency of the system. Protection units that are placed in series with the load consume some amount of power and, therefore, generate heat. These items may or may not be significant, but they should be considered. Prepare a complete life-cycle cost analysis of the protection methods proposed. The study may reveal that the long-term operating expense of one system outweighs the lower initial purchase price of another.

Dollars and sense

The amount of money a broadcaster is willing to spend on protection from utility company disturbances is generally a function of how much money is available in the engineering budget and how much the station has to lose. Spending $25,000 for system-wide protection for a major-market station, where spot rates can run into the hundreds or thousands of dollars, is easily justifiable. At small- or medium-market stations, however, justification is not so easy.

The susceptibility of electronic equipment to failure because of disturbances on the ac power line has been studied by many organizations. The best benchmark study conducted to date was done by the Naval Facilities Engineering Command (Washington, DC). The far-reaching program, conducted between 1968 and 1978 by Lt. Thomas Key, identified three distinct categories of recurring disturbances on utility company power systems. As shown in Table 1, the duration of the disturbance, not the magnitude of the voltage, determines the classification

DEFINITION	TYPE 1 Transient and oscillatory overvoltage	TYPE 2 Momentary undervoltage or overvoltage	TYPE 3 Power outage
CAUSES	Lightning, power network switching, operation of other loads	Power system faults, large load changes, utility company equipment malfunctions	Power system faults, unacceptable load changes, utility equipment malfunctions
THRESHOLD*	200 to 400% of rated RMS voltage or higher (peak instantaneous above or below rated RMS)	Below 80-85% and above 110% of rated RMS voltage	Below 80-85% of rated RMS voltage
DURATION	Spikes 0.5 to 200μs wide and oscillatory up to 16.7ms at frequencies of 200Hz-5kHz and higher	From 4 to 60 cycles depending on type of power system distribution equipment	From 2 to 60 seconds if correction is automatic; from 15 minutes to 4 hours if manual

*The approximate limits beyond which the disturbance is considered to be harmful to the load equipment.

Table 1. The types of voltage disturbances identified in the Key report.

Type of disturbance	UPS system and standby generator	UPS system	Secondary spot network[1]	Secondary selective network[2]	Motor-generator	Shielded isolation XFMR	Suppressors, filters, lightning arrestors	Solid-state line-voltage regulator
1	All source transients No load transients	All source transients No load transients	None	None	All source transients No load transients	Most source transients No load transients	Most transients	Most source transients No load transients
2	All	All	None	Most	Most	None	None	Some depending on response time of system
3	All	All outages shorter than battery supply discharge time	Most	Most	Only brown-outs	None	None	Only brown-outs

NOTES:
1. Dual power feeder network.
2. A dual power feeder network using a solid-state switch to select which line is fed to the load.

Table 2. The types of systemwide protection equipment available to broadcasters and the ac line abnormalities that each approach is capable of handling.

in Key's categories.

The study found that most equipment (in this case computer) failure caused by ac line disturbances occurred during periods of bad weather. In fact, according to a report on the findings of the study, the incidence of thunderstorms in a given area may be used in predicting future failures.

The type of power transmission system used by the utility company also was found to have an effect on the number of disturbances observed on power company lines. For example, an analysis of utility system problems in Washington, DC, Norfolk, VA and Charleston, SC, showed that underground power distribution systems experienced only a third as many failures as overhead lines in the same areas.

Tables 2 and 3 show the various options available to station engineers to protect sensitive broadcast equipment from ac line disturbances, and the approximate costs for that protection. Because each installation is unique, conduct a thorough investigation of the station's needs before any equipment is purchased.

Assessing the lightning hazard

As identified by Key in his Naval Facilities study, the extent of lightning activity in a given area has a significant effect on the probability of equipment failure because of transient activity. The threat of a lightning strike to a given facility is a function of several factors associated with the installation and its location. These factors include the geographical location, type and character of the facility, plant size and the character of the lightning strike itself.

The Keraunic number of a geographical location represents the likelihood of lightning activity in the given area. Figure 3 shows the Isokeraunic map of the United States, which estimates the number of lightning days per year in various areas of the country. There is an average of 30 storm days per year across the continental US. This number, however, does not fully describe the lightning threat because many individual lightning strikes occur within a single storm.

The structural character of a particular facility will have a significant effect on the lightning threat to equipment operation. Higher structures tend to collect -- and even trigger -- lightning strikes. And because storm clouds tend to travel at specific heights above the earth, conductive structures in mountainous areas will trigger lightning activity more readily.

The plant exposure factor is a function of the size of the facility and the Isokeraunic rating of the particular area. The larger the physical size of an installation, the more likely it will be hit by lightning during a storm. It also follows that the longer a transmission line (ac or RF), the more lightning strikes it will likely receive.

The character of a lightning strike covers a wide range of voltage, current and rise-time parameters. Making an accurate estimation of the damaging potential of a strike is a difficult proposition. A direct hit to a utility power line will cause a high-voltage, high-current wave to travel away from the point of the hit in both directions along the power line. The waveshape is sawtooth with a rise time measured in microseconds or nanoseconds. The pulse travels at nearly the speed of light until it encounters a

Basis of comparison[1]	UPS system and standby generator	UPS system	Dual power feeders	Motor-generator	Shielded isolation XFMR	Suppressors, filters, Lightning arrestors	Solid-state line-voltage regulator
Installation and equipment costs	$1500 to $2000 per kVA	$1100 to $1500 per kVA	Installation cost will vary greatly depending on site	$250 to $400 per kVA	$50 to $150 per kVA	$1 to $10 per kVA	$250 to $280 per kVA
Maintenance costs	$2000 to $4000 per year	$1100 to $3000 per year	None	Less than $1000 per year	None	None	Less than $1000 per year
Operating efficiency[2]	80-85%	80-85%	100%	80-90%	Up to 98%	100%	90-98%

NOTES:
1. A power conditioning system rated for approximately 25kVA is assumed.
2. Efficiency applies to the ac power conditioning equipment only. Losses in environment support systems are not taken into account.

Table 3. The approximate cost of systemwide protection equipment installation and operation. Equipment prices can vary considerably.

significant change in line impedance, at which point it is reflected back down the line in the direction it came from. This action creates a standing wave containing the combined voltages of the two original pulses. A high energy wave of this type can reach sufficient potential to arc over to another parallel line, a distance of about 8 feet on a local feeder (typically 12kV) power pole. Not surprisingly, such energy can have devastating effects on electronic equipment.

The relative frequency of power problems observed by users appears to be seasonal in nature. As shown in Figure 4, the majority of problems are noted during the months of June, July and August. The likely cause of these high problem rates can be traced to increased thunderstorm activity and heavy, unpredictable air conditioning loads during the summer months.

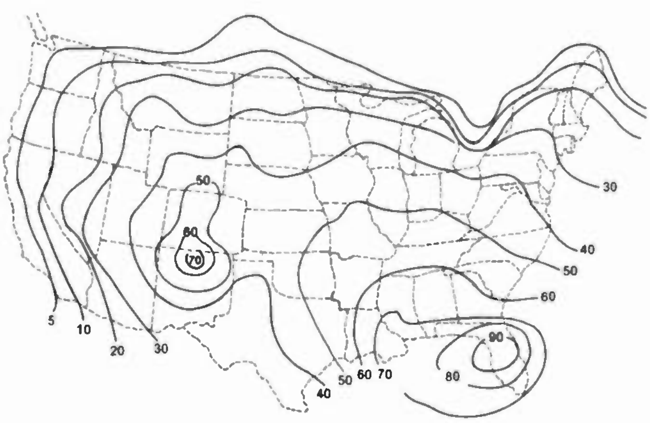


Figure 3. The Isokeraunic map of the United States, showing the approximate number of lightning days per year.

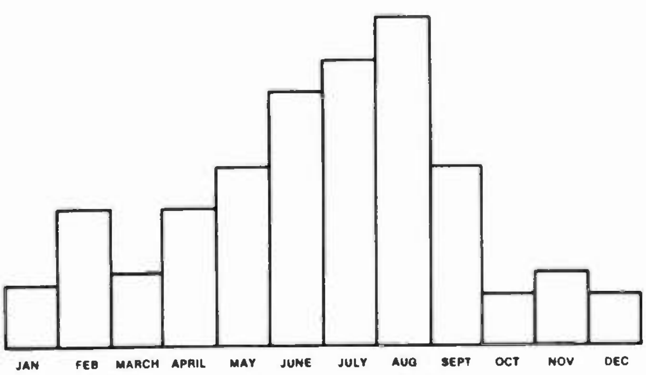


Figure 4. The relative frequency of power problems in the United States classified by month. (Reference 1.)

FACILITY PROTECTION METHODS

Proper grounding of equipment and structures at a broadcast plant is basic to protection against ac line disturbances. This applies whether the source of the disturbance is lightning, power system switching activities or faults in the distribution network. Regardless of which protection approach used, all protective devices and systems require a solid, low resistance earth ground to operate properly. Grounding is important at both the studio and the transmitter plant. Probably the greatest challenge to proper grounding lies with a mountain-top transmitter location.

The grounding arrangement for a remote location grounded-tower (FM or TV) transmitter plant generally follows the guidelines shown in Figure 5. The tower and guy wires are grounded using copper-clad ground rods measuring 10 ft. in length. The antenna is bounded to the tower and the transmission line is bonded to the tower at the point it leaves the structure and begins the horizontal run into the transmitter building. Before entering the structure, the line is bonded to a ground rod through a connecting cable. The transmitter itself is grounded to the ac power distribution system ground, which -- in turn -- is bonded to a ground rod where the utility feed enters the building. The design goal of this arrangement is to strip all incoming lines of damaging overvoltages before they enter the facility. One or more lightning rods are mounted at the top of the tower structure. The rods extend at least 10 ft. above the highest part of the antenna assembly.

The grounding configuration shown in Figure 5, although commonly found at many installations, has built-in problems that can make it impossible to provide adequate transient protection to equipment at the site. Consider the Figure 5 example again.

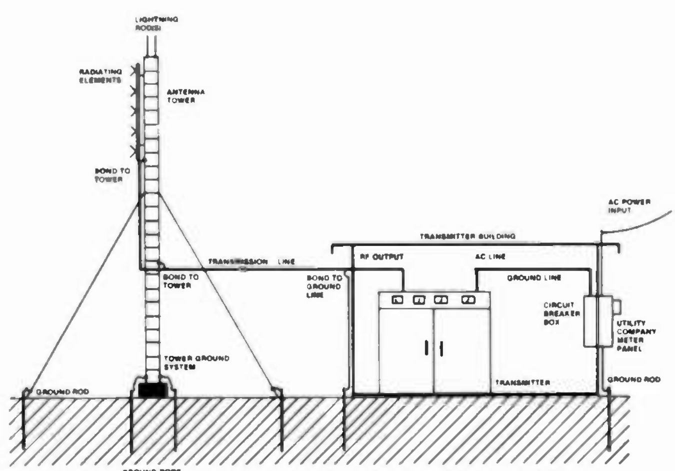


Figure 5. The typical, but not ideal, grounding arrangement for a transmission facility using a grounded tower. A better configuration involves the use of a bulkhead panel through which all cables into and out of the equipment building pass.

To equipment inside the transmitter building, two grounds actually exist -- the utility company ground and the antenna ground. One ground will have a lower resistance to earth, and one will have a lower inductance in the connecting cables or copper strap from the equipment to the ground system. Assume that a transient overvoltage enters the utility company meter panel from the ac service lines. The overvoltage is clamped by a protection device at the meter panel and the current surge is directed to ground. But which ground, the utility ground or the antenna ground?

The utility ground will surely have a lower inductance to the current surge than the antenna ground, however, the antenna will probably exhibit a lower resistance to ground than the utility side of the circuit. The surge current will, therefore, be divided between the two grounds, placing the transmission equipment in series with the surge suppressor and the antenna ground system. Given a transient of sufficient potential, damage will result to the transmission equipment.

Transients generated on the antenna side because of a lightning discharge are no less troublesome. The tower is a conductor, and any conductor is also an inductor. A typical 150-foot self-supporting tower may exhibit as much as 40µH inductance. During a fast-rise-time lightning strike, an instantaneous voltage drop of 360kV between the top of the tower and the base is not unlikely. If the coax shield is bonded to the tower 15 feet above the earth (as shown in Figure 5), 10% of the tower voltage drop (36kV) will exit at that point during a strike. The only way to ensure that damaging voltages are stripped off of all incoming cables (coax, ac power and telephone lines) is to use a bulkhead entrance panel.

Bulkhead panel

The concept of the bulkhead is simple: establish one reference point to which all cables entering the equipment building are grounded and to which all transient suppression devices are mounted. Figure 6 shows the basic approach.

The bulkhead panel size depends on the spacing, number and size of the coaxial lines entering the building through the plate. The panel should be made of copper or brass. Do not use steel, unless it is stainless steel (18-8 type or equivalent). To provide a weatherproof point for mounting transient suppression devices, the bulkhead can be modified to provide a subpanel, as shown in Figure 7. The subpanel is attached so it protrudes through an opening in the wall and creates a secondary plate on which suppressors are mounted and grounded. To handle the currents that may be experienced during a lightning strike, the bottom-most subpanel flange (which joins the subpanel to the main bulkhead) must have a total surface contact area of at least 0.75 square-inch per transient suppressor.

Because the bulkhead panel will carry significant current during a lightning strike or ac line disturbance, it must be constructed of heavy material. The recommended material is 1/8-inch C110 (solid copper) 1/2 hard. This type of copper stock weighs nearly 5 1/2 pounds per square foot and sells for about $2.25 per pound. Installing a bulkhead is an expensive job, but one that will pay dividends for the life of the facility. Use 18-8 stainless steel mounting hardware to secure the subpanel to the bulkhead.

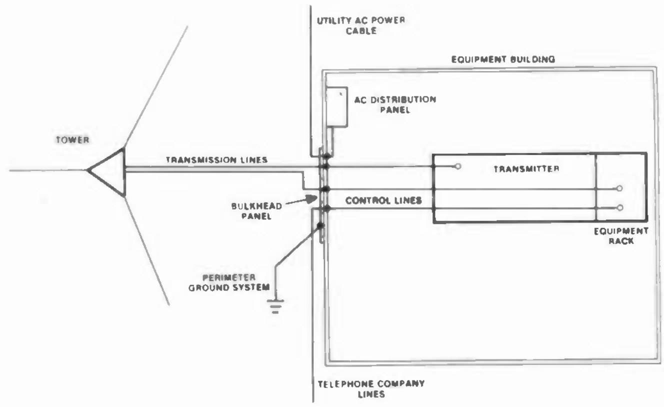


Figure 6. The basic design of a bulkhead panel for a transmission facility. (Reference 2.)

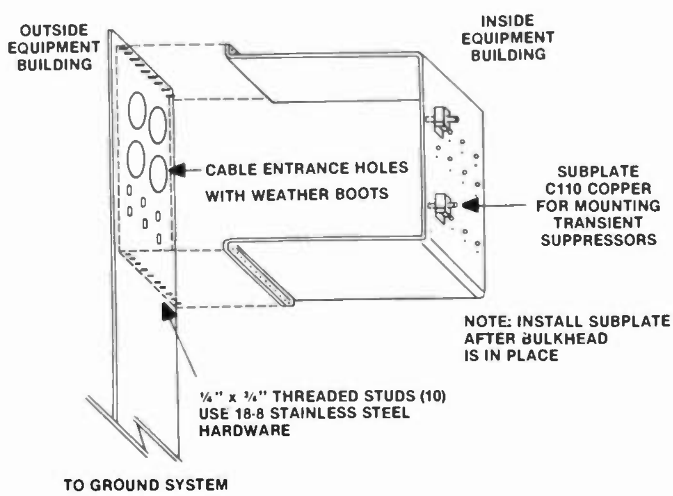


Figure 7. The addition of a subpanel to a bulkhead as a means of providing a mounting surface for transient suppression components that is not exposed to outside elements. (Reference 2.)

Because the bulkhead establishes the central grounding point for all equipment within the building, it must be tied to a low resistance (and low inductance) perimeter ground system. Ideally, the bulkhead panel will extend down the side of the building and tie into the perimeter ground below grade level. This will result in the lowest resistance and inductance to earth ground.

Checklist for proper grounding

Follow a logical series of steps to ensure that the needed protection can be achieved at the broadcast plant:

1. Install a bulkhead panel to provide mechanical support, electrical grounding and lightning protection for coaxial cables entering the equipment building.

2. Install an internal ground bus using #2 or larger solid copper wire (at transmission facilities use copper strap 3-inches or wider). Form a "star" grounding system. At larger installations, form a "star-of-stars" configuration. Do not allow ground loops to exit in the internal ground bus.

Connect the following items to the building internal ground bus: chassis racks and cabinets of all equipment, and all auxiliary equipment (chargers, switchboards, conduits, metal raceway and cable trays).

3. Install a tower earth ground array by driving ground rods and laying radials as required to achieve a low earth ground resistance at the site.

4. Connect outside metal structures to the earth ground array (towers, metal fences, metal buildings and guy anchor points).

5. Connect the power line ground to the array. Be certain to follow local electrical code.

6. Connect the bulkhead to the ground array through a low inductance and low resistance bond.

Do not use soldered-only connections outside the equipment building. Crimped, brazed and exothermic (Caldwelded) connections are preferred. To make a proper bond, all metals surfaces must be clean, any finish removed to bare metal and surface preparation compound applied. Protect all connections from moisture by appropriate means (sealing compound and heat-sink tubing).

The ac wiring system

Most transient disturbances that a facility will experience enter the plant through the utility company ac power line. Effective transient suppression, therefore, begins with proper installation of the ac power system wiring. Arrange with the local utility to have a separate transformer feed your facility. This request may cost more initially, however, it will reduce the chances of transient disturbances from nearby operations affecting your equipment. Do not allow noisy loads to be placed on the broadcast facility power line. Devices such as arc-welders, heavy electrical motors, elevators and other large loads can create an electrical environment that is prone to equipment malfunctions.

Insist that all ac wiring within the broadcast facility be performed by an experienced electrical contractor, and always fully within the local electrical code. Figure 8 shows a typical service entrance panel, with the neutral line from the utility company tied to ground and to the ground rod at the meter panel. Where permitted by the local code, this should be the only point at which neutral is tied to ground in the ac distribution system.

Figure 9 shows a 3-phase power distribution panel. Note that the neutral and ground connections are kept separate. Most ac distribution panels give the electrical contractor the ability to lift the neutral from ground by removing a shorting screw in the breaker panel chassis. Insulate the neutral lines from the cabinet. Bond the ground wires to the cabinet for safety. Always run a separate insulated green wire for ground. Never rely on conduit or other mechanical structures to provide ground to electrical panels or equipment.

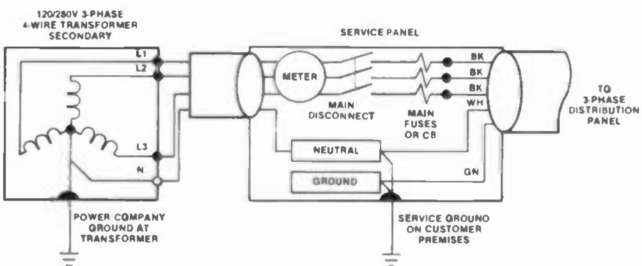


Figure 8. The recommended connection method for a 3-phase utility company service panel. (Reference 3. Copyright 1986 Howard G. Mullinack.)

A single-phase power distribution panel is shown in Figure 10. Note that neutral is insulated from ground and that the insulated green ground wires are bonded to the panel chassis.

If allowed by the local electrical code, do not use metal conduit in cable runs between distribution panels and wall socket outlets. Use PVC pipe, Romex or jacketed cable. This will eliminate the possibility of electrical contact with other conduit and metal structures in the facility. If metal pipe is required by code, be certain it does not contact other conduit runs in the system.

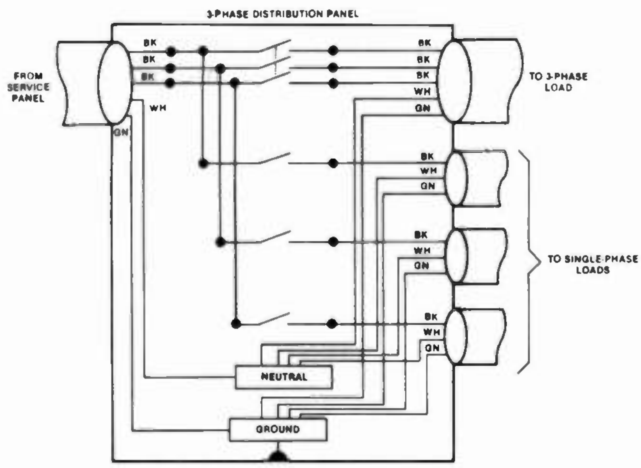


Figure 9. Arrangement of the neutral and green-wire ground system for a 3-phase ac distribution panel. (Reference 3. Copyright 1986 Howard G. Mullinack.)

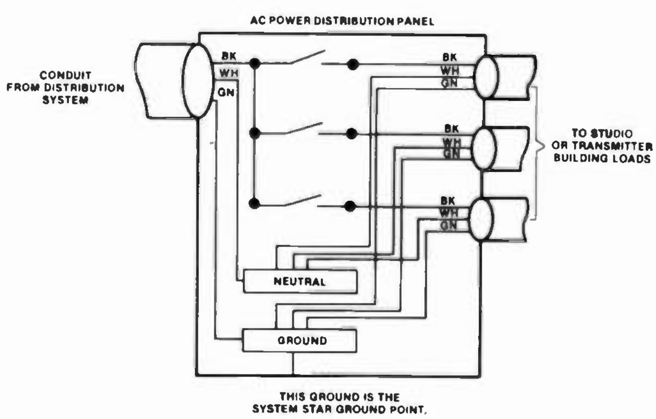


Figure 10. Arrangement of the neutral and green-wire ground system for a single-phase ac distribution panel. (Reference 3. Copyright 1986 Howard G. Mullinack.)

The wiring of equipment racks for electrical power must also be carefully planned. Bond adjacent racks together with bolts and clean the contacting surfaces by sanding down to bare metal. Mount equipment in the rack using the normal mounting rack screws. If located in a high RF field, clean the rack rails and equipment panel connection points to ensure a good electrical bond. Connect the circuit ground from each piece of equipment to a system ground strap that is insulated from the rack. Use standard 3 prong grounding ac plugs. Do not defeat the safety ground connection. Insulate the ac conduit (if used) from the rack by using insulated bushings between the conduit and the receptacle box. Bond the rack to the green wire ground at the receptacle box.

Before implementing any type of ac power system changes, review the entire project with a qualified electrical contractor.

Discrete device protection

There are two basic ways to protect a broadcast facility from transient disturbances: the systems approach or the discrete device approach. Table 2 outlines the major alternatives available for the systems approach to transient suppression:

1. UPS (uninterruptable power supply) system and standby generator,
2. UPS stand-alone system,
3. Secondary ac spot network,
4. Secondary selective ac network,
5. Motor-generator unit,
6. Shielded isolation transformer, and
7. Solid-state line-voltage regulator.

The systems approach offers the advantages of protection engineered to a particular application and need and (usually) high-level factory support during system design and installation. The systems approach, however, has the drawback of higher cost. Many facilities cannot justify spending $5,000 or more for a sophisticated protection system. For such installations, the only alternative is to consider the application of discrete protection devices at critical points in the ac power system.

In the business of transient protection -- like anything else -- you get what you pay for. Discrete devices are less expensive and usually provide a reduced level of protection, compared to a sophisticated systems approach. It is unrealistic for a user to expect a group of discrete transient suppressors to do the job of a much more expensive systems design. However, properly applied discrete devices can provide a level of protection sufficient to prevent damage to equipment from all but the most serious transient disturbances. The key to achieving this level of performance lies in understanding and properly applying discrete protection devices.

The performance of discrete transient suppression components available to the broadcast engineer has greatly improved in the last 10 years. Transient suppression technology has come a long way from the days of spark gaps and resistor-capacitor (RC) snubbers.

The wide variety of new devices available at reasonable prices make tight control over unwanted voltage excursions possible, and allow the complicated electronic equipment being manufactured today to work as intended. Much of the credit for transient suppression work goes to the computer industry, which has been dealing with the problem for more than two decades.

Types of devices

Transient suppression hardware can be divided into three general categories: ac filters, crowbar devices and voltage-clamping components.

The simplest type of ac power line filter is a capacitor placed across the voltage source. The impedance of the capacitor forms a voltage divider with the impedance of the source, resulting in the attenuation of high-frequency transients. This simple approach has definite limitations in spike suppression capability, and may introduce unwanted resonances with inductive components in the ac power distribution system.

The addition of a series resistance will reduce the undesireable resonant effects, but will also reduce the capacitor's effectiveness in attenuating a transient disturbance.

Crowbar devices include gas tubes (also known as spark-gaps or gas-gaps) and semiconductor-based active crowbar protection circuits. Although these devices and circuits have the capability of shunting a substantial amount of transient energy, they are subject to power-follow problems.

Once a gas tube or active crowbar protection circuit has fired, the normal line voltage, as well as the transient voltage, will be shunted to ground. This power-follow current may open protective fuses or circuit breakers if a means of extinguishing the crowbar clamp is not provided.

Voltage-clamping devices are not subject to the power-follow problems common in crowbar systems. Clamping devices include selenium cells, zener diodes and varistors of various types.

Zener diodes, using improved silicon rectifier technology, provide an effective voltage clamp for the protection of sensitive electronic circuitry from transient disturbances. Power dissipation for zener units is usually, however, somewhat limited (compared with other suppression methods).

Selenium cells and varistors -- although different in construction -- act in similar ways on a circuit exposed to a transient overvoltage. Figure 11 illustrates the variable nonlinear impedance exhibited by a voltage-clamping device, and how these components are capable of reducing transient overvoltages in a particular circuit.

The voltage divider network established by the source impedance (Z_S), and the clamping device impedance (Z_C), acts to attenuate voltage excursions at the load. It should be understood that the transient suppressor depends upon the source impedance to aid the clamping effect. A protection device cannot be fully effective in a circuit that exhibits a low source impedance, because the voltage divider ratio is proportionately reduced.

A typical voltage-vs.-current curve for a voltage clamping device is shown in Figure 12. When the device is exposed to a high voltage transient, the impedance of the component changes from a high standby value to a low conduction value, thereby clamping the voltage at a specified level.

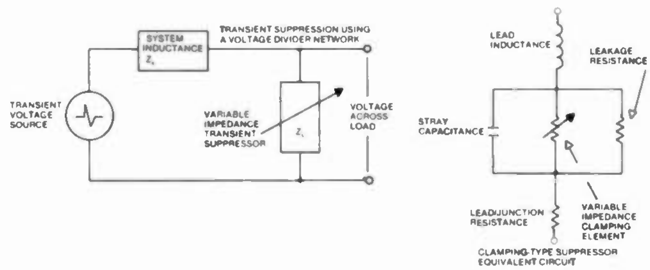


Figure 11. The mechanics of transient suppression using a voltage clamping device.

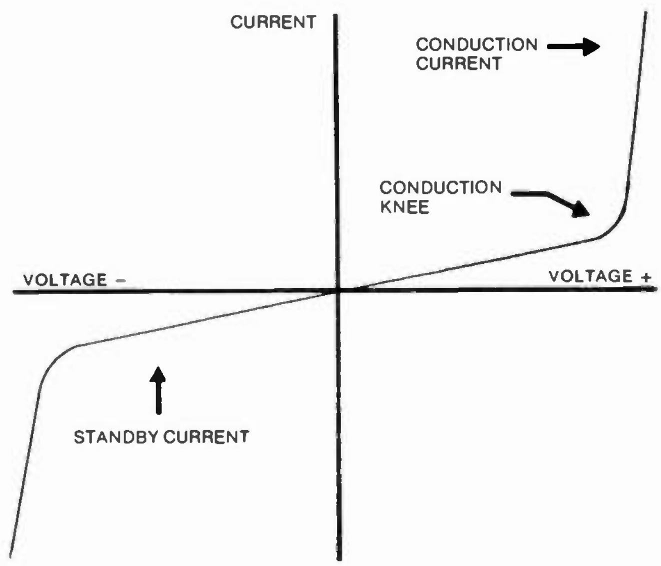


Figure 12. The voltage-vs.-current curve for a typical bipolar voltage clamping device. The component is designed to be essentially invisible in the circuit until the applied positive or negative potential reaches or exceeds the conduction knee of the device.

Selecting a protection device

Selecting a transient suppression device for a particular application is a complicated procedure that must take into account the following items:

1. The steady-state working voltage,

including normal tolerances.
2. The transient energy to which the device is likely to be exposed.
3. The voltage clamping characteristics required in the application.
4. Circuit protection devices (such as fuses or circuit breakers) present in the system.
5. The consequences of protection device failure in a short-circuit mode.
6. The sensitivity of the load equipment to transient disturbances.

Most transient suppression equipment manufacturers offer detailed application handbooks. Consult such reference data whenever you plan to use a protection device. The specifications and ratings of suppression components are not necessarily interchangeable from one manufacturer to another.

Carefully weigh the addition of transient suppression devices to a piece of equipment or ac power distribution system. Make allowances for operation of the circuit under all anticipated conditions.

Plan ahead

There is nothing magical about effective transient suppression. Disturbance on the ac line can be suppressed if the protection method applied has been carefully designed and properly installed. Whether the protection method your station chooses involves a systems approach or discrete devices at key points in the broadcast plant, the time and money spent incorporating protection into your facility will pay for itself in short order.

Transient disturbances are a fact of life. The power quality in the United States is not improving. With increased loading and diminished reserves in some areas, it is, in fact, becoming worse. Broadcasters will have to pay the bill for transient disturbances one way or the other: either for protection hardware or equipment maintenance after the fact. The choice is yours.

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Acknowledgment:

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BROADCAST AM SYNCHRONOUS TRANSMISSION

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ABSTRACT

The use of nearby AM transmitters with the same frequency and program is widely practiced throughout the world. Most installations, however, are for equal power transmitters in mono. The problems of a high power transmitter and a low power transmitter for fill-in both with stereo modulation have not been widely analyzed or reported.

The version of synchronous AM currently being experimentally tried by several U.S.A. stations is different from that used in other parts of the world. The FCC has experimentally permitted some stations to broadcast from a low powered station within the coverage area of their high powered transmitter site. Several problems exist with this arrangement.

INTRODUCTION

Present attention in AM has focused on many of the real technical difficulties of medium wave broadcast. Several things can be done to improve broadcast AM....standardize pre-emphasis/de-emphasis for radio receivers, reduce unwanted skywave, reduce adjacent channel interference, use AM stereo, and use of synchronous sites. This paper will focus on the use of AM synchronous to help broadcasters.

The use of nearby AM transmitters with the same frequency and program is widely practiced throughout the world. Most installations, however, are for equal power transmitters in mono. The problems of a high power transmitter and a low power transmitter for fill-in both with stereo modulation have not been widely analyzed or reported.

The problems of synchronous mono operation have been described by CCIR, EBU and in older literature in the U.S.A. How synchronous AM stereo performs remains largely unknown. In an effort to shed some light on this problem a study was made and results of this study compiled for an understanding of the problem.

We must take note that FCC licensed synchronized AM transmissions have not been permitted, except under experimental conditions. But synchronized AM transmission is receiving considerable attention in the U.S.A. Actions by the FCC seem to indicate that if all goes well in some field trials, this technique will be authorized. What you may be able to do by filling "holes" in your coverage is exciting to think about!

U.S.A. STYLE

The contemplated FCC version of synchronous AM is different from applications which use equal power transmitters in mono. In the U.S.A., the primary transmission signal completely overlaps a secondary transmission site. The potential for an interference zone is, therefore, much higher. Since most people don't understand this zone of interference, it is worthwhile discussing the problems surrounding a low power transmitter surrounded by a much higher powered transmitter. How one implements a system for maximum benefit isn't completely known yet. However, many known principles can be applied and the results of the field trials analyzed. To get a grasp of this potential, let's look at a system and some of its parts.

SIGNALS

To understand what is basically happening, an analogy may help. What we know is that signals from a single tower create concentric circles of equal strength. This is likened very much to dropping a large pebble into a quiet pool of water, then dropping a smaller pebble some distance away. As expected, the waves caused by either pebble near the entry point are not affected by the other. Additionally, none of the waves around the large pebble are affected very much either. However, at a short distance from the small pebble entrance an interference zone will be created. This pebble analog is identical to the electrical case we'll examine.

The secondary site is within the coverage of the primary site. Reception on a receiver is like that of "co-channel interference," but the program audio material is identical. Another description of reception is like that of ground wave and sky wave cancellation of signals received from a single station. This type of signal structure is like a transmission line. The primary site signal is at one end and the secondary site signal at the other. Early experiments and recent tests indicate that if the two sites are exactly synchronized, then the "best" coverage can be obtained. To RF carrier synchronize sites it is possible to use:

* Satellite references

* WWV signals

* Ultra-stable oscillators

* Comparison to other RF sources in areas such as FM transmitters, Loran, etc.

Each of these methods have certain advantages, shortcomings, and associated cost. Another method of RF locking, audio distribution, and remote control can be rolled together as one system by using presently available STL equipment. A practical link system for virtually any synchronous site can be designed. A block diagram is shown in Figure 1. This system has excellent features and uses currently available equipment with some simple techniques to get optimum results.

SYNC EQUIPMENT

The equipment for the synchronous site is much like that of the main site. But let me describe some ideas to help minimize overall costs:

Site Choice

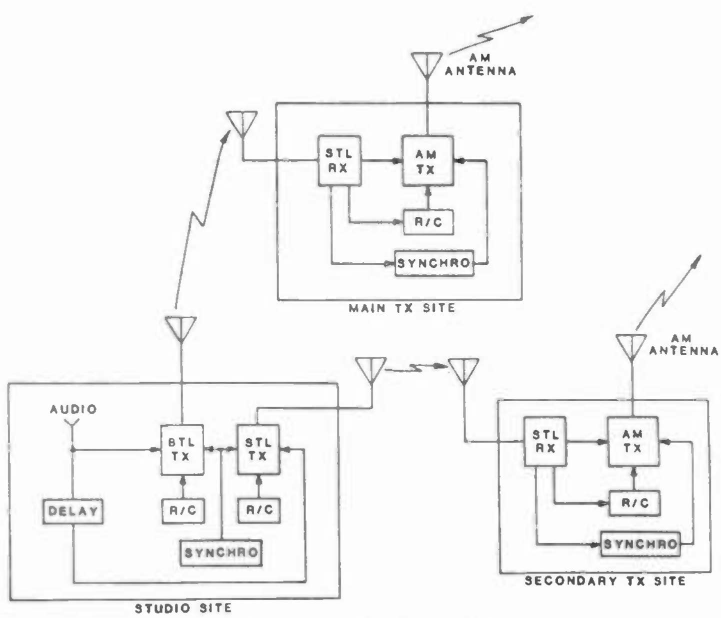
* Use someone else's radio tower(s) (diplexed operation)
* Use any other suitable tower/radiator

Antenna

* Use a very short tower and topload
* Use a portable erectable antenna

Building

* Use someone else's building
* Use shelterized building



TYPICAL BLOCK DIAGRAM
FOR SYNCHRONOUS OPERATION

Figure 1

The other equipment you'll need includes:

* Transmitter (likely 1% to 10% of your main)
* Remote control (ATS remote?)
* Audio processing (minimal)
* RF sync lock
* STL transmitter/receiver
* Digital audio delay

All of this equipment is readily available and can be optimized for best reception.

USING SYNCHRONOUS

Several broadcasters have an interest in using a synchronous site(s). The reasons are similar; the causes are different:

* Obstructions: natural terrain variations or manmade structures
* Pattern variations: nulls in antenna pattern
* Interference: Co-channel and adjacent channels usually skywave

All of these causes are candidates for synchronous site solutions. Tall buildings and mountains provide obstacles which make coverage with a single site impossible. By strategically locating a low power synchronous site(s), coverage can be recovered to the FCC theoretical limits. Some stations have the interesting problem of being able to see their antenna towers but being unable to clearly hear it! Yes - even 50KW stations!

STEREO

AM stereo adds complicating factors to synchronous operation. Both AM stereo systems (Kahn and C-QUAM) use low frequency pilot tones to signal the transmission of stereo. Stereo thus adds the potential need for synchronizing the pilot tones for each site. Since only C-QUAM receivers are available in numbers, that system deserves some attention.

Some thought has been given to out-phasing the C-QUAM pilots. Special properties of the C-QUAM decoder chips and their ability to switch to mono makes this an unique opportunity for good reception with stereo receivers. Field tests will be run soon to determine the most effective mode of pilot phase.

Most AM stereo generators generate the pilot tone reference from the same crystal reference as the carrier. Thus, any synchronizing signal that replaces the crystal reference can synchronize the RF carrier <u>and</u> pilot tone.

PREDICTING THE FIELD

Predicting the contours of signal ratio is necessary to evaluate the impact of a synchronous site. Beginning with these assumptions - omnidirectional, flat terrain, and ideal conductor - then we know that the field strength (volts per meter) from an antenna will vary inversely with the distance from the antenna. This field strength pattern is shown in Figure 2 for two transmitter sites. In Figure 2, the transmitters are 10:1 different in power.

Examining Figure 2 we can note the following:

* The signal from the high power transmitter is nearly constant over the region around the low powered transmitter.
* The signal strength from the low powered transmitter drops very quickly from the antenna, thus minimizing excessive coverage.
* The signal strength from the high powered transmitter diminishes slowly and dominates the coverage area.
* As one travels along a path from the primary site (#1) to the secondary site (#2), the signal from #1 diminishes and at some point will become equal in strength to site #2 signal.

RATIOS

Calculations will show that the contours of constant signal ratio (signal strength from #1 compared to the signal strength from #2) are circles. This is shown in Figure 3.

If you would like to calculate what may be applicable for your situation, you may start by using some simple approximations. First, we'll assume flat terrain, omni, and perfect conductivity. Let us define:

* $P1$ = Power of Transmitter #1 (Watts)
* $V1$ = Signal Strength of Transmitter #1 (V/m)

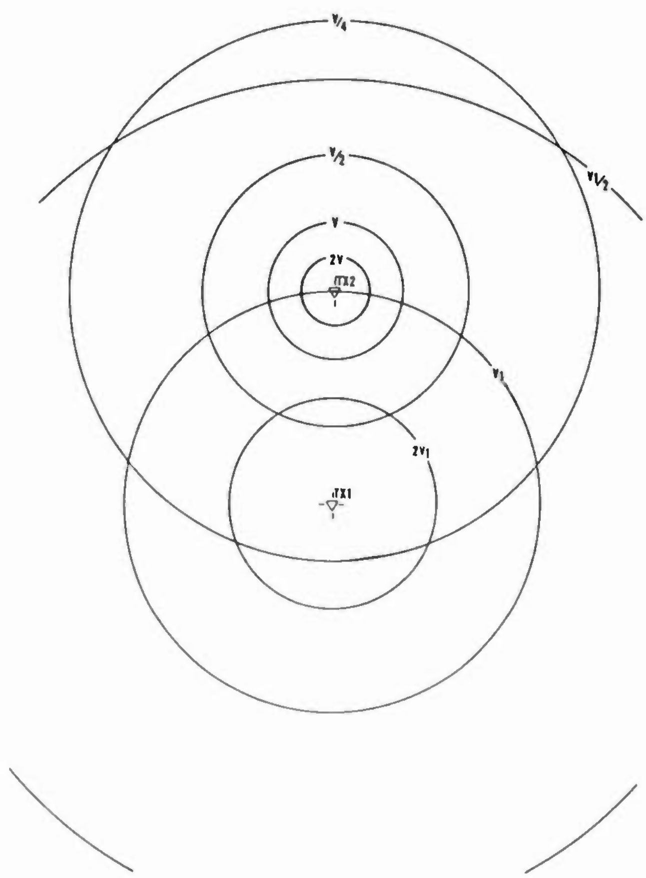


Figure 2

* P2 = Power of Transmitter #2 (Watts)
* V2 = Signal Strength of Transmitter #2 (V/m)
* D = Distance Separated (miles)

Then you can calculate the circles of equal strength by using:

$$R = D \quad \frac{k}{1-k} + \left(\frac{k}{1-k}\right)^2 \qquad \text{(miles)}$$

$$Y = -\frac{(kD)}{1-k} \qquad \text{(miles)}$$

$$k = \left(\frac{V2}{V1}\right)^2 \left(\frac{P1}{P2}\right)$$

Where:

R = Radius of circle with center at point Y.
Y = Distance from Site #1 on line drawn through Site #1 and Site #2.
k = Ratios of Interest

By using these formulas, some insight into the coverage can be gained. But finite ground conductivity will change these results. What may happen is that the area "behind" the secondary site may not have a strong primary signal and thus no losses to listeners will occur. A thorough study by your consultant can answer these questions.

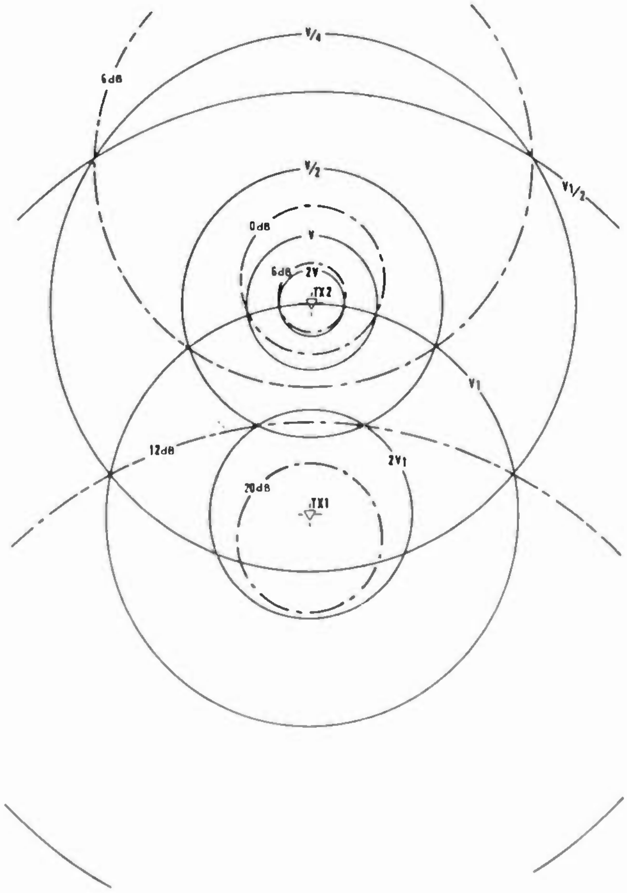


Figure 3

TEST RESULTS

Several stations are using AM synchronous sites under experimental FCC licenses. The problems they are solving are real. Other stations have applied for an experimental license or are planning to do so. Broadcasters using 500W, even to 50KW transmitter are interested. Daytimers and clear channel broadcasters alike are interested in synchronous operation.

Some test results have demonstrated the basic idea...covering an area not well covered before with no additional interference to other stations. The "zones of interference" problem exists, but strategic locationing and power selecting has minimized listener losses. If a system is designed without RF synchronizing, then some problems may exist. Without complete phase locking (e.g. 0.1 Hz difference), the degradation may be heard.

By traveling along highways and using auto radios and portable radios, the following qualitative data was gathered. Because the main and secondary site were not PLL synchronous, the RF signal strength was varying. This variation was used to calculate the ratio of signals from the primary and secondary sites. These results were noted:

Max. Field	Min. Field	Ratio	Comments
1.7 mV/m	1.25 mV/m	16.0 dB	Some noticeable degradation, listenable, not objectionable.
1.4 mV/m	0.45 mV/m	5.8 dB	Considerable audible degradation.
1.0 mV/m	0.21 mV/m	3.5 dB	Considerable audible degradation. Other stations audible during minimums.

For signal ratios of 20 dB or greater, no degradations were noted on auto or portable radios. Because this data is for a weak signal, AGC of the receiver became part of the observed results.

The audio quality in the zone of interference is best evaluated by listening to recordings. However, the following descriptions for non PLL sync sites are applicable:

Degradation Level	Notes
Field Strength Varies ± 5%	No noted deterioration
Onset of Degradation	Pulse of degradation every 1-5 seconds
Maximum	Volume pumps, skywave heard on minimum, distortion maximum on maximum field strength. Audibility best at low signal strength, noise bursts, cancellations every 5 seconds.

CONCLUSION

Since the secondary site is within the coverage of the primary site, reception on a receiver is like that of "co-channel interference" but the program audio material is identical. What will reception be like? We know reception can be made very good and potential problems minimized. As you might guess, if the primary signal is a hundredfold more powerful (20 dB) than the secondary, virtually no problems are expected. Similarly, if the secondary signal overwhelms the primary, then few problems exist. It is in the zone where the two RF signals are nearly equal that problems may be expected. This "zone of interference" needs to be understood and calculated so the benefits of synchronous AM are not overstated.

The most important part of the "zone of interference" is where the two RF signals are of equal strength. Experiments and calculations indicate the following ratios are important:

Signal Power Ratio	Importance
1:1 (0 dB)	Contour of maximum interference
1:1 to 4:1 (0 dB to 6 dB)	Zone of some interference
4:1 to 100:1 (6dB to 20 dB)	Zone of little interference
100:1 and greater (20 dB)	Zone of virtually no interference

Synchronous transmitters can be expected to create many of the effects commonly associated with skywave/groundwave interference and selective fading. Path length and attenuation differenced under skywave conditions create disturbances of carrier phase and comb filtering of the sidebands. Quadrature distortion on envelope detectors results, as well as periodic dips in frequency response. Whenever the sites are PLL'ed, these problems are minimized. Reception will be nearly "normal" and coverage losses virtually nil.

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ANALYTICAL METHODS TO IMPROVE DIRECTIONAL ANTENNA EFFICIENCY

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ABSTRACT

For some years now, microcomputers have been assisting the RF engineer in the design and maintenance of directional antenna systems. With the increase in popularity of the lap-top computer, field project time can be expedited and a more thorough analysis performed on-site. It will be shown that through the use of computers better understanding of array operation and possible increase in efficiency may be realized.

INTRODUCTION

Any array has two major factors considered in its design: pattern size and pattern shape. Pattern size, referred to as the RMS value is given by

$$RMS=\sqrt{\frac{E_1^{\,2}+E_2^{\,2}+E_3^{\,2}...etc.}{N}} \qquad (1)$$

where E_1, E_2, etc.=inverse distance field at E_1 azimuth, etc.
N=number of radials

Pattern shape, determined by the so-called Cosine Law defines the value of inverse distance field on a given azimuth. FCC Rules and Regulation 73.150 stipulates a 1 ohm loss be placed at the current loop (point of maximum current) on each element. The overall value of RMS will, of course, be reduced due to the dissipation of power in this 1 ohm loss. The "power lost" may be determined by first determining the loop current in each element:

$$\text{Loop In } \frac{K\ F_N}{C\ (1-\cos G_N)} \qquad (2A)$$

or

$$\frac{K\ F_N}{C\ (\cos(TL_N)-\cos(TL_N+G_N))} \qquad (2B)$$

Where:
- K=no-loss pattern constant (mi or km)
- C=37.256479 for K given in miles
  59.958491 for K given in km
- G_N=height of Nth element
- F_N=field ratio of Nth element
- TL_N=top loading (degrees) of Nth element

Equation 2A applies to regular towers,2B applies to the top-loaded case

Now, by Ohm's Law, the power dissipation is given by:

$$P_{L_N}=I^2R \qquad (3)$$

- P_L=Power lost in each element
- I_N=Loop I given in Eq. 2
- R=loss resistance, usually 1 ohm

It can be seen that by rationalizing Ohm's Law, the short tower case which has an inherently higher loop current will dissipate a greater power loss than its taller counterpart. Other factors may serve to worsen this problem.

MODING

Let us examine a two tower array with the following parameters:

Height	Phasing	Field Ratio	Spacing
90	0	1	0
90	+95	.556	90

(FIG 1)

The MASTER® computer program generates the following data using formulae given a 1 ohm loss at the current loop:

Tower	Loop I	Power	Loop op R	Loop opX
1	4.67	946	43.4	+j88.8
2	2.60	54	7.95	+j42.5

(FIG 2)

It is apparant that one tower is radiating most of the power. While some may consider this an advantage in terms of bandwidth, the stability of the pattern is questionable. Given the fixed constants of height, spacing and orientation, we can "mode" the pattern by simply swapping field ratios, thus:

Tower	Loop I	Power	Loop op R	Loop opX
1	2.60	396	58.8	+j115.6
2	4.67	604	27.7	+j66.2

(FIG 3)

The process of "moding" may be applied to arrays synthesized by the multiplication of two-tower arrays. Each pair, one by one may be inverted or "moded." A general form of moding may be applied to <u>any</u> array; however, for dog-leg or unequally spaced arrays it would be necessary to relocate one or more towers. Program 1 is a simplified approach to moding and reflects the alternative array parameters generated in the FCC's RADIAT computer program. The reader is advised to avoid moding of unequal height tower pairs. Such practice would alter the vertical (conical radiation) pattern and negate the original design limits.

PATTERN ADJUSTMENTS

Prior to making impedance measurements and adjusting networks, it is important to consider whether the pattern is actually correct, that is, as specified on your CP. A popular method of initial array tune-up involves talking in the protected radials to zero field and then "calibrating" the sample system either algebraically or by trimming sample lines. The array is then readjusted to the theoretical parameters using the "calibration" previously described. This method, which replaces the "TWO TOWER TEST"(1), may be utilized to insure that an array is adjusted as close to the theoretical parameters as possible.

In order to "prove" our adjustments, we need to obtain field intensity measurements from certain directions from the array. Once the data is gathered and analyzed, the inverse distance fields may be determined. The actual IDF is usually determined by data gathered at monitor points less than 2 miles from the array. When gathering data close to an antenna(s), the influence of the magnetic (H) field(s) will not permit the actual electric field intensity to be displayed. A method of proximity correction may be utilized in the analysis of this data in order to more accurately reflect the true IDF's. Let us examine the PROXY® computer program output for the parameters shown.

Tower	Field Ratio	Phase	Spacing	Azimuth
1 (ref)	1	0	0	0
2	1	-149.5	90	178

Radial (deg)	Distance (mi)	Correction factor
280	.72	1.242
280	1.12	1.145
300	.21	.777
300	.59	.905
300	1.45	.959

(FIG 4)

Before using proximity correction, several important points must be made. First, we make the (sometimes dangerous) assumption that the array is adjusted to the theoretical parameters. It was previously described how to make certain your array is as close to the theoretical parameters as possible using a talk-in approach aided by computer. Second, we must make certain that the radials are plotted on the topo map with the reference tower as origin. This is especially important with wide-spaced arrays. For some arrays, the reference point is at the center of the array. The ALTARR® program can select any of the towers as the reference tower, thus:

Tower	Spacing	Azimuth
1	72	340
2	17.5	250
3	72	160

=

Tower	Spacing	Azimuth
1	74.1	353.66
2	0	0
3	74.1	146.34

(FIG 5)

BANDWIDTH

Once we are certain that the pattern is in proper adjustment, we may proceed to consider the bandwidth. A systematic approach is recommended when overall improvements to the system are to be made:

1. Measure the driving point impedance at each tower. Be sure to retrim the phasor for the correct parameters if the bridge insertion effect is noticeable. Make certain the OIB is placed <u>before</u> the toroid sample, if used.

2. Measure the phase shift of each network. This may be accomplished by using the antenna moniotr and two toroid sam-

ple transformers. Alternately, an oscilloscope and two current sample probes may be used. If neither of the above are readily available, you can measure the individual component values and plug them into the TANALYZ program (program 2).

3. Check the accuracy of your thermocouple common point and base ammeters against a known standard. Now perform I^2R power radiated calculated. How well do your powers add up? If there is an appreciable power loss, check the following:

a) Be certain all radials are properly bonded to the base ring.
b) Be certain the base ring is properly bonded to the ATU.
c) Be certain a good connection is made to the straps running between towers and the strap from the towers to the phasor and transmitter.
d) Check base insulator, with a megger.
e) Check transmission and sample lines with a megger.
f) Immediately after sign-off, check for capacitor heating. Check coil contacts for signs of burning.
g) Have your tower man check for loose bonding between tower sections.

4. With all of this information in hand, one can now attempt redesign or readjustment of the networks with the following guidelines:

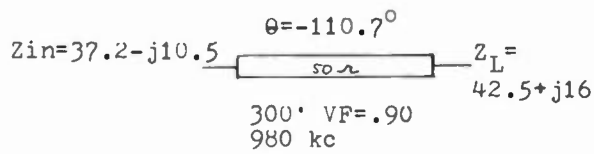
a) Keep all network phase shifts close to 90°.
b) Keep all L/C ratios as low as possible.
c) Do not use L/C combinations where they are not necessary.
d) When using the Ohm's Law shunt power divider, do not use a power control for the reference tower. Connect the reference tower directly at the common point matching network.

5. Keep in mind that the current phase shift of a transmission line is equal to:

θ (electrical degrees) = θ (physical length in degrees) x velocity factor

(4)

Equation 4 will hold true only when VSWR= 1.0:1. In the case of a mismatched line, we may use the LINEZO computer program (program 3) to determine the actual current phase shift.



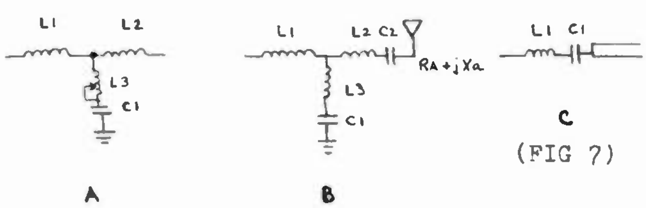
(FIG 6)

Note that in the above example, the electrical length of the line is -119.5, while its actual phase shift equals -110.63 degrees.

6. A good directional array will exhibit constant pattern bandwidth at the sideband frequencies. By this I mean the radiation pattern will not severely change shape at the sideband frequencies. An easy test method for pattern bandwidth would be to temporarily substitute a variable frequency oscillator into the transmitter's exciter and, using only sufficient power to drive the antenna monitor, move the carrier frequency ± 10 kc and observe the phase and loop ratio on each tower. Any drastic variations require attention.

7. A system diagram should be made noting all load impedances, line impedances. You can now begin to see the overall system phasing as actually adjusted. The basis for your redesign and/or adjustment will come from this data sheet.

8. We mentioned to avoid high L/C ratios in a previous statement. Often times, a coil is used as a fine-tuner or to make a capacitor "variable." Let's look at three typical series L/C circuits:



(FIG 7)

In Fig 7-A L3 is used to make C1 variable. In most cases, if X_C were made equal to 1.5 times the required X_C and X_L were made equal to required X_C, an acceptable ratio would be obtained. In Fig 7-B, the output arm of the network actually contains three reactances. In some instances, L_2 and/or C_2 could be eliminated. In Fig 7-C we see a familiar front panel phase control. L_1 and C_1 are normally tuned to resonance. The phase shift at any other adjustment is:

$$\Theta = \text{ARCTAN } \frac{X}{R} \qquad (5)$$

There are times where a high L/C ratio can be of advantage. Let us consider the following tabulation of common point impedance.

Fo	Zcp
990	55 +j15
1000	50 ± 0
1010	45 -j15

(FIG 8)

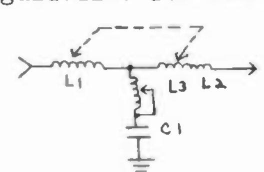
Let's add a series resonant circuit consisting of a .00021 uf capacitor and a 118.5 uh coil. The transformed common point is:

Fo	Zcp
990	55 ± 0
1000	50 ± 0
1010	45 ± 0

(FIG 8A)

The series network components have +j744 and -j744 impedances at carrier. Ordinarily, we try not to use these high ratio networks where element phasing and power division occur. This is due to the fact that a minor component variation could place the array parameters out of tolerance. Also, such a network could produce drastically different results at the sideband frequencies since resonance occurs only at F_C.

9. All networks should be designed to properly match the transmission lines. A Tee phase shifter serves as a simple guideline for set-up:



(FIG 9)

First L_1 and L_2 are set to the center of their range. A 50 ohm non-inductive resistor (cantenna) is placed as the load. A carrier-frequency oscillator serves as the RF source. An impedance bridge is placed at the input terminals and L3 is adjusted to present a 50 ± j0 impedance.

The ATU networks may be adjusted using an OIB in the "hot" mode. Remember, the stand-alone (self) base impedance is <u>not</u> the driving point impedance, so an ATU network <u>cannot</u> be adjusted using towers individually. The ATU networks serve two key functions: First, to provide the proper phasing and second, to provide a proper termination to the transmission lines. By examining the overall diagram you have just made, it can be seen how to uniformly distribute the system phasing so that the networks can be set to ± 90° or as close thereto as practical. Network redesign may be attributed to the following reasons:

1) Poor base impedance predictions in the original design.
2) Change in transmission lines such as from RG-17/u to heliax. Velocity factor changes from 60% to 90%.
3) Additions to towers such as STL, RPU, ANT, etc.
4) Changing from steel to Phyllistran guy wires.
5) Replacement of ground system.
6) Addition of lightning "top hats" creating a top loading effect (and attendant change in vertical radiation characteristics).

CONCLUSION

It is hoped that sufficient insight and reason has been shown as to how certain considerations can improve directional antenna efficiency and bandwidth. A personal computer can provide assistance in the analysis of network and radiator characteristics so as to improve the efficiency of a directional antenna array.

(1) The Two Tower Test by Robert Jones Broadcast Engineering Magazine

PROGRAM 1

```
10 CLEAR
20 PI=3.14159265358979#
30 RADIAN=PI/180
40 CLS:KEY OFF
50 PRINT"[ALT] ALTERNATIVE ARRAY COMPUTER PROGRAM"
60 PRINT "COPYRIGHT @1986 BY THOMAS G.OSENKOWSKY (203)775-3060
70 PRINT
80 INPUT "NUMBER OF TOWERS";T
90 PRINT :PRINT
100 FOR A=1 TO T
110 PRINT"RATIO TOWER";A;:INPUT F(A)
120 PRINT "PHASE TOWER";A;:INPUT P(A)
130 PRINT "SPACING TOWER";A;:INPUT S(A)
140 PRINT "AZIMUTH TOWER";A;:INPUT A(A)
150 PRINT
160 NEXT A
170 ORI$="ORIGINAL ARRAY "
180 AL$="ALTERNATIVE ARRAY "
190 CLS
200 PRINT ORI$:PRINT
210 PRINT"TWR   SPACING   PHASING   FIELD   AZIMUTH"
220 FOR A=1 TO T
230 PRINT USING "##    ###.##    ####.##    #.###    ###.##";A;S(A);P(A);F(A);A(A)
240 NEXT A:PRINT
250 PRINT AL$:PRINT
260 PRINT"TWR   SPACING   PHASING   FIELD   AZIMUTH"
270 A=0:FOR B=1 TO T
280 IF SGN(P(B))=-1 THEN PP(B)=ABS(P(B))
290 IF SGN(P(B))=1 THEN PP(B)=-(P(B))
300 A=A-1
310 AA(B)=A(B)+180
320 IF AA(B))=360 THEN AA(B)=AA(B)-360:GOTO 320
330 IF S(B)=0 THEN AA(B)=0
340 PRINT USING "##    ###.##    ####.##    #.###    ###.##";ABS(A);S(B);PP(B);F(B
);AA(B)
350 NEXT B
360 END
```

PROGRAM 2

```
10 CLS:KEY OFF
20 PRINT"[TANALYZ] COPYRIGHT @1986 BY THOMAS G. OSENKOWSKY (203)775-3060"
30 PRINT:PRINT
40 PRINT"THIS PROGRAM ALLOWS THE USER TO ENTER VALUES FOR A REAL-WORLD"
50 PRINT"TEE NETWORK WHICH MAY BE SEEING A MISMATCHED CONDITION."
60 PRINT:PRINT
70 PRINT"***IN THE CASE OF A REACTIVE LOAD, ANY SERIES COIL OR CAPACITOR"
80 PRINT"MUST BE TAKEN INTO ACCOUNT. AN EXAMPLE OF THIS WOULD BE AN ANTENNA"
90 PRINT"WHOSE OPERATING IMPEDANCE HAS AN INDUCTIVE REACTANCE. TYPICALLY, A"
100 PRINT"CAPACITOR IS INSERTED BETWEEN THE ANTENNA AND ATU OUTPUT ARM."
110 PRINT"THIS CAPACITOR WOULD HAVE AN Xc VALUE CLOSE TO THAT OF THE XL VALUE"
120 PRINT"OF THE ANTENNA LOAD."
130 PRINT"THE REACTANCE OF THIS CAPACITOR MUST BE TAKEN INTO ACCOUNT FOR EACH"
140 PRINT"ANALYSIS PERFORMED. IF YOU ARE MAKING MORE THAN ONE ANALYSIS, SAY"
150 PRINT"FOR SIDEBANDS, KEEP IN MIND Xc AND XL CHANGE FOR EACH FREQUENCY."
160 PRINT"ALSO, IF YOU READ IMPEDANCES OFF A BRIDGE, BE SURE TO SCALE EACH"
170 PRINT"REACTANCE MEASURED TO ITS RESPECTIVE FREQUENCY.IF REDESIGNING"
180 PRINT"A NETWORK, BE SURE TO KEEP XL/Xc RATIOS AS LOW AS POSSIBLE.***"
190 PRINT:PRINT
200 PRINT
210 INPUT"ARE YOU ENTERING DATA IN REACTANCE (X) OR COMPONENT VALUES (C)";FD$
220 PRINT
```

```
230 IF FD$="" THEN 210
240 IF FD$="X" THEN 270
250 IF FD$="C" THEN 1060
260 PRINT
270 PRINT"**********OBSERVE SIGNS OF EACH ARM AND LOAD R & X**********"
280 PRINT
290 X1=0:X2=0:X3=0:X4=0:X5=0
300 INPUT"VALUE OF X1 (INPUT ARM)";X1
310 INPUT"DOES OUTPUT ARM CONSIST OF A SINGLE (S) COMPONENT OR TWO (T)";OA$
320 IF OA$="" THEN OA$="S"
330 INPUT"VALUE OF X2 (OUTPUT ARM)";X2
340 IF OA$="S" THEN 360
350 INPUT"VALUE OF X4 (LOAD ARM)";X4
360 INPUT"DOES SHUNT ARM CONSIST OF A SINGLE (S) COMPONENT OR TWO (T)";OS$
370 IF OS$="" THEN OS$="S"
380 INPUT"VALUE OF X3 (SHUNT ARM)";X3
390 IF OS$="S" THEN 420
400 INPUT"VALUE OF X5 (SHUNT TUNE ARM)";X5
410 X3=X3+X5
420 IF OA$="S" THEN 440
430 X2=X2+X4
440 IF R$="Y" THEN R=ZZ:X=YY:GOTO 470
450 INPUT"LOAD RESISTANCE";R
460 INPUT"LOAD REACTANCE";X
470 ZZ=R:YY=X
480 PRINT
490 RR=(X3^2*R)
500 RI=RR/(R^2+((X+X2+X3))^2)
510 XJ=X3*(R^2+(X+X2)*(X+X2+X3))
520 XE=XJ/(R^2+(X+X2+X3)^2)
530 XI=X1+XE
540 PRINT USING"INPUT RESISTANCE= ###.## ";RI
550 PRINT USING"INPUT REACTANCE= ####.## ";XI
560 M=R/(X+X2+X3)
570 C=ATN(M)
580 C=C*(180/3.1416159#)
590 IF C>0 AND X3<0 THEN C=C-180
600 IF C<0 AND X3>0 THEN C=C+180
610 PRINT USING"PHASE SHIFT OF NETWORK= ####.## ";C
620 F=2.734*C
630 F1=C/3.28
640 T1=1.2*F*.0174532925#
650 A=1000!
660 B=.0005
670 C=1
680 T=TAN(T1)
690 R=RI/50
700 H=XI/50
710 D1=((1-(T*H))^2)+((R*T)^2)
720 RO=(R*(1+T^2))*C/D1
730 XO=((T*(1-H^2-(H*T)-R^2))+H)*C/D1
740 D2=RO^2+(2*RO)+XO^2+1
750 R1=SQR(((RO^2 + XO^2-1)^2)+(4*XO^2))/D2
760 S=(1+R1)/(1-R1)
770 SW=INT(A*(S+B))/A
780 CC=INT(A*(RO+B))/A
790 DD=ABS(INT(A*(XO+B))/A)
800 IF XO<0 THEN A$=" - " ELSE A$=" + "
810 PRINT USING"SWR FOR 50 OHM LINE = ###.## ";SW
820 PRINT"NORMALIZED Z=";CC;A$;"J ";DD
830 PRINT
840 PRINT"THE FOLLOWING SECTION IS ONLY FOR GUIDELINE PURPOSES. IT ASKS IF"
850 PRINT"THE USER WISHES TO FIND THE LOAD RESISTANCE FOR A GIVEN (DESIRED)"
860 PRINT"INPUT RESISTANE i.e. 50 OHMS. USABLE RESULTS ARE NOT ALWAYS OBTAINED"
870 PRINT"USING THIS SECTION."
880 INPUT"DO YOU WISH TO SOLVE FOR DESIRED INPUT RESISTANCE (Y/N)";Z$
890 IF Z$="Y" THEN 900 ELSE 1030
900 PRINT
```

```
910 INPUT"DESIRED INPUT RESISTANCE";RI
920 V=SQR(((X3^4)-(4*(RI^2))*((X+X2+X3)^2)))
930 Q=X3^2-V
940 R=Q/(2*RI)
950 PRINT"ANTENNA RESISTANCE FOR DESIRED VALUE OF Rin";R
960 X7=-X2-X3-SQR((X3^2*R/RI)-R^2)
970 PRINT"LOAD REACTANCE FOR DESIRED VALUE OF Rin ";X7
980 X4=RI*(YY+X2)
990 X5=SQR(ZZ*(RI)*((X+X2)^2)+(RI)*(ZZ^3)-(ZZ^2*RI^2))
1000 X6=X4+X5
1010 X7=X6/(ZZ-RI)
1020 PRINT"VALUE OF SHUNT ARM (X3) FOR DESIRED Rin ";X7
1030 INPUT"DO YOU WISH ANOTHER RUN (Y/N)";R$
1040 IF R$="Y" THEN 240
1050 KEY ON:END
1060 PI=2*3.14159265#
1070 PRINT"ANALYZE TEE NETWORK PARAMETERS"
1080 PRINT
1090 PRINT"**********INPUT VALUES IN MICROFARADS AND MICROHENRIES**********"
1100 X1=0:X2=0:X3=0:X4=0:X5=0
1110 PRINT
1120 IF R$="Y" THEN 1150
1130 INPUT"OPERATING FREQUENCY (KHZ)";FK
1140 FK=FK/1000
1150 INPUT"VALUE OF X1 (INPUT ARM)";X1
1160 IF X1>1 THEN 1190
1170 X1=1/(PI*FK*X1):X1=-X1
1180 GOTO 1200
1190 X1=PI*FK*X1
1200 INPUT"DOES OUTPUT ARM CONSIST OF A SINGLE (S) COMPONENT OR TWO (T)";OA$
1210 IF OA$="" THEN OA$="S"
1220 INPUT"VALUE OF X2 (OUTPUT ARM)";X2
1230 IF OA$="S" THEN 1290
1240 INPUT"VALUE OF X4 (LOAD ARM)";X4
1250 IF X4>1 THEN 1280
1260 X4=1/(PI*FK*X4):X4=-X4
1270 IF X4<1 THEN 1290
1280 X4=PI*FK*X4
1290 IF X2>1 THEN 1320
1300 X2=1/(PI*FK*X2):X2=-X2
1310 GOTO 1330
1320 X2=PI*FK*X2
1330 IF OA$="S" THEN 1350
1340 X2=X2+X4
1350 INPUT"DOES SHUNT ARM CONSIST OF A SINGLE (S) COMPONENT OR TWO (T)";OS$
1360 IF OS$="" THEN OS$="S"
1370 INPUT"VALUE OF X3 (SHUNT ARM)";X3
1380 IF OS$="S" THEN 1440
1390 INPUT"VALUE OF X5 (SHUNT TUNE ARM)";X5
1400 IF X5>1 THEN 1430
1410 X5=1/(PI*FK*X5):X5=-X5
1420 GOTO 1440
1430 X5=PI*FK*X5
1440 IF X3>1 THEN 1470
1450 X3=1/(PI*FK*X3):X3=-X3
1460 GOTO 1490
1470 X3=PI*FK*X3
1480 IF OS$="S" THEN 1500
1490 X3=X3+X5
1500 R=ZZ:X=YY
1510 IF R$="Y" THEN 1540
1520 INPUT"LOAD RESISTANCE";R
1530 INPUT"LOAD REACTANCE";X
1540 GOTO 470
```

PROGRAM 3

```
10 REM LINEZO: XMSN LINE PROGRAM
20 INPUT"LINE CHARACTERISTIC IMPEDANCE";ZO
30 INPUT"VELOCITY FACTOR";VO
40 IF VO=0 THEN VO=1
50 INPUT"LINE LENGTH IN FEET";L
60 INPUT"OPERATING FREQUENCY (MHZ)";FO
70 INPUT"LOAD RESISTANCE";RO
80 INPUT"LOAD REACTANCE";XO
90 REM COMPUTE IMPEDANCE
100 BO=L*FO*6.2831853#/984:BO=BO/VO
110 K1=ZO+XO*COS(BO)/SIN(BO):K2=RO/ZO
120 K3=-RO*COS(BO)/SIN(BO)
130 K4=XO/ZO-COS(BO)/SIN(BO)
140 RI=(K1*K2+K3*K4)/(K2*K2+K4*K4)
150 XI=(K2*K3-K1*K4)/(K2*K2+K4*K4)
160 REM COMPUTING VSWR
170 R=(RO+.00001)/ZO:X=(XO+.00001)/ZO
180 K1=(R-1):K2=(R+1)
190 K3=K1*K2+X*X:K4=K2*X-K1*X
200 K4=SQR(K3*K3+K4*K4)/(K2*K2+X*X)
210 SW=(1+K4)/(1-K4):SW=ABS(SW)
220 CLS:PRINT"TRANSMISSION LINE DATA":PRINT
230 PRINT USING"ZO= ###.#  ";ZO
240 PRINT USING "VELOCITY FACTOR= #.##  ";VO
250 PRINT USING"LENGTH IN FEET ####.##  ";L
260 PRINT USING"FREQUENCY (MHZ) ##.##  ";FO
270 BO=-BO*360/6.2831853#
280 PRINT USING"PHASE DELAY OF LINE #####.##  ";BO
290 PRINT"LOAD RESISTANCE";RO
300 PRINT"LOAD REACTANCE";XO
310 PRINT"INPUT RESISTANCE";RI
320 PRINT"INPUT REACTANCE";XI
330 PRINT USING"LINE VSWR  ###.##  ";SW
340 L=L*(3.14159/180)
350 X3=ZO/SIN(L)
360 X2=Z/TAN(L)-X3
370 M=R/(X+X2+X3)
380 C=ATN(M)
390 C=C*(180/3.14159)
400 IF C>0 THEN C=C-180
410 PRINT"CURRENT PHASE SHIFT OF LINE=";C
```

HIGHWAY INFORMATION RADIO SYSTEMS — WHAT THEY DO, HOW THEY WORK

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PRESENTATION SUMMARY

Highway Information Radio Systems have much in common with AM broadcasting. The technology is closely related to the low power authorizations granted daytime broadcasters, and to synchronized AM broadcasting. The growing utilization of highway radio systems makes it appropriate for the broadcast engineer to understanding them. Prospective users often seek direction from local broadcasters.

Applications of highway information radio systems in which the motorist uses his conventional AM radio receiver offer a significant advantage over roadside signs. The applications are highly varied and the technology can be confusing because of the variety of possible approaches. Numerous systems are currently operational throughout the US. This presentation reviews the history, the technology and several applications of the various forms of highway information radio.

INTRODUCTION

Highway information radio systems are described by different names. The Federal Communications Commission calls the licensed systems Travelers' Information Stations (TIS), the Federal Highway Administration calls them Highway Advisory Radio (HAR). Additional confustion results from the several different Rules under which the necessary AM signal can be transmitted. Each has advantages and disadvantages, depending upon the application, operational requirements and site considerations. Similar systems are also used for drive-in theaters, drive-in churches ... even for drive in-banks and car washes ... and for assistance to the hearing impaired and language translation.

This discussion is limited to systems in which the motorist utilizes his conventional and unaltered AM radio for reception of information while driving along a roadway. The several ways of producing an AM signal for the motorist should be of interest to broadcast engineers because of the similarity of technology, and because prospective users often turn to broadcasters for initial direction about highway information radio systems.

HISTORIC APPLICATIONS

First, let's review a cross section of highway information radio systems. This will provide an understanding of some of the different problems.

The George Washington Bridge

Nothing's new under the sun! A highway information radio system on the George Washington Bridge provided directions for motorists Eastbound to the 1939 New York's World's Fair. A published paper describing the system discusses a revolutionary new German wire recorder used for the repeating message! This pioneering installation was by a well-known entrepeneuring broadcast engineer of the time, William S. Halstead. He subsequently developed a unique triaxial cable for multifunction highway radio systems which we will look at it later. Bill is now in retirement in Woodland Hills, CA.

Los Angeles International Airport System

Historically, perhaps the next installation was on the Century Boulevrd approach road to the Los Angeles International Airport in 1972. The City of Los Angeles obtained an experimental license from the FCC to operate this system on 530kHz. The system was conceived to provide the approaching motorist information about the complex traffic pattern around the airport, where the airline terminals are located, where parking is currently available, and accidents or other situations which might delay traffic. LAX is a busy and complex airport where passing your destination means another time consuming trip around the loop.

This system used two broadcast zones ... we will show how to create these in a minute. The first provided the motorist a welcome to LAX and general orientation information. He proceeds over a short overpass and finds himself in another message zone which provides specific real-time information about traffic and parking.

Real-time highway information radio requires the timely collection and dissemination of information. To do this, LAX and their contractor, LocRad Inc., established a studio in one of the terminal-area buildings. Here information is gathered from the airport police and from CCTV cameras at key locations. After a period of time, tape carts covering about all possible situations had been inventoried. The operator on duty in the studio picks the appropriate tape cart for insert into the familiar cart machine. When necessary, live information is easily presented.

It is fair to state that this LAX system is the forerunner of the 1978 authorization of Travelers Information Stations in Part 90.242 of the Commission's Rules. The LAX system was transferred from an experimental license to the TIS Rules and has been in continuous operation for 15 years! LocRad Inc. continues to be the contract operator for the Airport.

Hershey Park

Hershey Park is a theme park at HERCO, the Hershey Chocolate complex in Eastern Pennsylvania. We installed a 540kHz highway radio system along one mile of the primary approach road. It provides information about current and future events and parking direction for Hershey Park and the adjacent Hershey Area which carries a heavy sports program. This system has been in continuous operation since 1981.

Raceway Radio

An interesting reversal of who's in motion is found in the systems installed at three raceways to provide spectators information via portable radio as the race cars whiz by. This noise level obliterates any hopes of hearing a public address system. In 1983 we installed an extensive system at the Summit Point Raceway in Summit Point WV, another at RoadAmerica in Elkhart Lake WI, and a third at the Oswego Speedway in Oswego NY. Many others have followed. The National Hot Rod Association has installed several.

Walt Disney World

Also in 1983, we installed systems on the entrance and exit roads for the opening of Walt Disney's EPCOT Center near Orlando, FL. Each of these is between three and four miles long. Disney personnel produce excellent tapes to advise the ariving visitor about parking, special attractions, hours of operation, various fare combinations, etc. This message is a visitor's first of Disney World, and it is a very good one. A exiting message invites a visit to the neighboring Magic Kingdom and provides information about it.

You may know that a similar system had been operating at Disney's Magic Kingdom for several years prior to our installation at EPCOT. It had been installed by the Disney organization. During the first week of operation of our EPCOT systems we received a call from our customer at Disney about a "problem" with our systems! His marketing people wanted to turn OFF the Magic Kingdom systems, they sounded so poorly in comparison to our new systems at EPCOT! Needless to say we promptly duplicated our EPCOT systems on the Magic Kingdom roads, and they have been in continuous operation since installation. These systems continue to win high marks from Disney personnel who have been kind in refering numerous inquiries to us.

Santa Anita Park

In the late summer of 1985 we installed a complex system at Santa Anita Park, a well-known race track in the Los Angeles area. It provides detailed race information to visitors within the track. This isn't a moving motorist system either, so we will pass by it quickly after mentioning it as another interesting application.

Plant Vogtle Nuclear Power Facility

About a year ago we installed a several mile system on the entrance road to the Plant Vogtle Nuclear Power Facility in Northeastern Georgia. This system provides information to employees as they begin their shift early in the morning, and later transitions into a visitor information system to give visitors a head-start before reaching the Visitor Center.

Atlantic City Expressway

We are now in the midst of the design and installation of a highway information radio system on the Eastern end of the Atlantic City Expressway, in southern New Jersey. This system is interesting bcause of the nature of this Expressway, and also because, when completed shortly, it will easily be the longest operating system. It will operate on 530kHz.

Centered about roughly Mile Post 25 (25 miles from Atlantic City) is a initial two-mile segment of two sequential messages. Since these messages do not require frequent change, they are stored and repeatedly read from digital storage units located with the transmitters in weatherproof enclosures along the Expressway.

As the Eastbound motorist clears the Pleasantville Toll Plaza another set of signs will alert him to the final 17 continuous miles of radio which will follow him into Atlantic City. First there he will hear a one-mile introductory message, then 15 miles of programming, and finally another one-mile terminating segment providing Atlantic City traffic directions. These segments will be programmed from studios in Atlantic City.

Others

We have reviewed a few applications to indicate the interesting scope of customers and applications for highway radio information systems. The two-dozen others we have installed include several other airports, systems along highway areas of severe weather and many visitor information systems.

THE APPLICABLE RULES

We see the FCC Rules as the place to begin consideration of how to implement a highway information radio system. There are three different Rules that may be applied, hence we have an area of prospective user confusion to clarify.

The appropriate choice will certainly relate directly to the user's needs and situation.

Unlicensed Highway Radio

One approach to achieve reception along the highway is to use the limited area AM broadcasting techniques which are allowed without an FCC license. Contrary to propular belief, there are conditions under which broadcasting is permitted IN the standard AM band without a license. Part 15 of the Rules contains these.

The purpose of Part 15 is to protect the licensed broadcast services from harmful interference created by a host of possible sources. Transmitter power, of and by itself, does not cause interference; it is the SIGNAL STRENGTH produced by the transmitter and antenna SYSTEM that causes interference. A high power transmitter with a very inefficient antenna might produce no interference. Conversly, a small low power transmitter operating on the frequency of a local broadcaster and employing an efficient antenna could provide severe interference. Part 15 therefore takes various approaches to place limitations upon the signal strength produced by a low power AM SYSTEM, rather than on transmitter power itself.

For operation in the AM broadcast band, Part 15 offers two options which we will review as they relate to highway information radio applications.

Restricted Radiation Devices

Part 15.7 allows the operation of unlicensed "restricted radiation devices," subject to certain reasonable restrictions about choice of operating frequency and resulting signal strength. The two basic restrictions are:

1) Choose a frequency so as not to interfere with a licensed broadcaster, and,

2) Produce a signal strength of no more than 15 microvolts per meter at a distance given by:

   157,000/F(kHz).

To choose to operate on a frequency not in local broadcast service is obvious, for one could not hope to overpower a broadcast station. We have never found a situation in which an acceptable, although perhaps not ideal, frequency could not be found.

The signal strength limitation of 15.7 is a bit more complex, but is important to understand. We think broadcast engineers will find it interesting. Just what signal strength is needed to be effective with a car receiver is not easy to put a number on. The answer depends upon the background noise on the frequency in the particular geographic location. It also depends upon the signal strength of broadcast stations the motorist is accustomed to receive. A highway radio system would like to produce a strong signal for easy acquisition and good intelligibility. Like broadcasters, we would like all the signal strength we can get!

A 15uv/m signal in the AM broadcast band is of no value. Note that the GREATEST distance at which 15uv/m is permitted is the LOWEST frequency ... 530kHz being the best available. There are several other solid arguments which also favor use of the lower broadcast frequencies. Finally, 530 is conveniently described to the user as the left end of the dial.

The 15uv/m position given by 157,000/F(kHz) is, from an analysis of the physics of the situation, that point in space where the Near and Far Fields are equal in magnitude. You may also know these as the Induction and Radiation Fields. Broadcasters work in the Far, or the Radiation Field. Highway information radio systems operate in the Near, or Induction Field. The important difference is that in the Induction Field the signal strength is a third power of distance, whereas in the Radiation Field it falls off linearly from the source. In the Induction Field, as one halves the distance to the source it is our good fortune that the signal strength increased by a factor of eight.

To demonstrate this important point with specific numbers, at 530kHz Part 15.7 allows 15uV/m at approximately 300 feet from the source. At 150 feet the measured E-field would be 120uV/m, at 75 feet 960uV/m, and at 35 feet approximately 8 millivolts! Now we are talking solid broadcast signal! A buried or suspended radiating cable alongside the road is an appropriate antenna for such a system. This is, in fact, the only form of "antenna" system which will comply with Part 15.7. The special coaxial cables used for this are often referred to as "leaky coax." There are several manufacturers whose cables take slightly different forms. The Disney, Hershey, Summit Point, Plant Vogtle and other systems are typical examples of highway information radio systems using "leaky coax."

The "100 milliwatt Rule"

Users often see the 100 milliwatt rule as an attractive solution to a limited area broadcasting requirment. Part 15.113 of the Rules allows the operation of a system using an antenna AND transmission line not in excess of 10 feet long, and a transmitter of input power to the final not over 100 milliwatts. Substantial recent publicity has been accorded the use of these systems for real estate sales. In the past we have done extensive development work on 100 milliwatt systems, and more recently have conducted extensive tests on the systems currently on the market.

We find the 100mw system yields dissapointing performance for many applications. In practice, reception on the typical car radio is limited to the range of 50 to 500 feet, depending on background noise, the quality of the installation, and antenna elevation. In this point source situation, unlike the line radiator of a cable along the road, the signal will be found only within a circle about the antenna. This system has potential application to small areas such as small parking lots and overlooks, but seldom meets the needs of highway radio applications. The system at Santa Anita Park in Los Angeles advantageously uses two 100mw systems to meet a portion of the customer's requirements. These are in addition to unlicensed Part 15.7 cable systems.

Traveler's Information Service

Travelers' Information Service (TIS) describes a licensed radio broadcasting technique for the dissemination of information to the motorist on his conventional AM car radio. This service was authorized in 1978 under Part 90.242 of the Rules. Since TIS is a licensed service, application must be made to the FCC, accompanied by appropriate technical data. These licenses are issued only to governmental organizations, and a TIS station is allowed to transmit only ...

"non commercial voice information pertaining to traffic and road conditions, traffic hazards and travel advisories, directions, availabaility of lodging, rest stops and service stations, and descriptions of local points of interest."

"It is not permissable to identify the commercial name of any business establishment whose services may be available within or outside the coverage area of a Travelers' Information Station. However, to facilitate announcements concerning departures/arrivals and parking areas at air, train and bus terminals, the trade name identification of carriers is permitted."

The requirement for speech bandwidth limitation in the TIS transmitter provides a reduction of sound quality in comparrison to accustomed broadcast reception. TIS stations may only be licensed on either 530kHz or 1610kHz. The Rules further require that the TIS station be located at least 15km outside the measured 0.5mV/mtr signal strength contour of the adjacent channel broadcaster. This means that one or both TIS frequenceis are not avaiable in some areas of the US. New York City and Long Island is an example of this. WLXI (250w-D, 540kHz) in Islip, at about the center of Long Island, wipes out the possibility of a 530kHz TIS station, and WWRL (5kw-U, DA-2, 1600kHz) whose towers are just across the river from Manhattan, takes out any 1610kHz TIS possibilities. We see this as somewhat ironic, for it is precisely such congested areas as New York City and Long Island that need the traffic advisory benefits of a Travelers' Information Service so badly.

To add to the user confusion, TIS stations may take either of two forms. The Rules allow the use of a radiating cable up to 1.9 miles long, or a vertical antenna no more than 15 meters above ground.

Vertical antenna systems seem attractive to most prospective users, perhaps because of the simplicity, economy, and success of CB radio systems, which are often seen as similar by the novice. Such similarity is not present, because of the much lower frequency of the AM broadcast band. The 15 meter TIS antenna height restriction severely limits antenna radiation efficiency. Also, in the AM band good antenna efficiency requires a ground plane under the antenna. By the time a reasonably efficient ground plane is installed, the materials, labor and occupied real estate may have become more costly than expected. In metropolitan areas a ground plane may be impossible to install.

We have a reasonable substitute in the form of a chemically treated ground rod, but it is costly and somewhat difficult to install. The rod is slightly less effective than an extensive ground radial system because the counterpoise effect is lost, but it appears to be less susceptible to fluctuations in ground conductivity and therefore is extremely stable in our experience thusfar.

For many TIS applications a vertical antenna has the disadvantage, compared to radiation cable, of 'point source' radiation characteristics. The reduction of signal strength with distance from the antenna makes it difficult for the motorist to acquire and identify the TIS signal at a distance down the road.

TIS Cable Systems

To add to the options avaialble, licensed TIS systems may take either of two forms. The Rules allow the use of cable, as with unlicensed Part 15.7 systems, up to 1.9 miles long, or the vertical antenna already described. A TIS cable has no operational advantage over a similar unlicensed system expect for a modest increase in allowable signal strength. Recall that unlicensed systems may utilize ANY available unused frequency in the AM band, while TIS systems are limited to 530 and 1610; and only then if either is locally available. We have pointed out that TIS is excluded from New York City and Long Island, yet we have been able to instal successful unlicensed 1150kHz cable systems on the three causeway approaches to Jones Beach on Long Island.

The signal strength produced by these systems is limited to 2mV/m at a distance of 60 meters from a radiating cable system, or 1.5 kilometers from a vertical antenna system. The approximate practical range of reception at these signal strengths is 20 meters from a radiating cable and 6 kilometers from a vertical antenna. In practice, transmiter power output to a radiating cable system is adjusted downward to provide coverage only to the lanes of traffic desired or the broadcast area desiread. The transmitter employed must carry an FCC Type Acceptance for this service. In any event, TPO is limited to 50 watts for cable systems and to 10 watts for vertical antenna systems.

Vertical vs: Cable

Vertical antennas provide coverage of circular areas. Cable systems are particularly suited to covering linear distances such a highways. Several cable systems can be installed in the same geographic area without mutual interference. The motorist can be provided segmented messages along the highway from one cable system to the next. The system at the Los Angeles International Airport uses two licensed TIS cables on 530kHz, and in recent years has been suplemented with A TIS vertical antenna located at a greater distance from the airport, also operating on 530kHz.

TRANSMISSION SYSTEMS

Unlicensed Radiating Cable Systems

We think of these systems in terms of basic one-mile "modules," where a mile of leaky coax is center fed from a transmitter of about 25 watts TPO with a toroidal power splitter on the output to drive two half-mile lengths of coax proceeding in opposite directions. Most suitable cables have a 50 ohm characteristic impedance. They are terminated in this impedance, hence are operated as terminated transmission lines for this application.

A system may be made with a message length much longer than one mile by adding linear RF amplifiers for each additional half-mile. An example of this is our systems at Disney.

Alternately, modules may be positioned end-to-end along the roadway to provide sequential messages. This allows an introductory message to be built upon in succeeding modules. An example is the system at LAX.

Triaxial Cable

There are at least four different leaky coax sources to choose from. The Halstead cable, manufactured by ComScope, is probably historically the first. As mentioned earlier, it is triaxial in nature, offering an inner 75 ohm coaxial system with an outer helix of a small copper ribbon. This helix works against the coax outer braid (actually a solid copper wrap) to provide the radiating cable. The inner coax was intended to provide for wideband high frequency information transmission. This cable is rather large, quite heavy, and requires a large benging radius which makes it difficult to work with. This is aggravated by the lack of a suitable connector.

The idea is to pick off and demodulate signal from the coax, using it to remodulate a broadcast band transmitter which uses the helix and coax outer conductor for radiation. This would allow a number of transmitters along the length of the cable. Alternately, the inner pair could provide another transmission means. For such dual-use applications, Mr. Halstead was granted a patent. We have not yet become aware of a dual-use requirement, but the growth of highway information radio systems may change this.

Slotted Cable

Andrews has adapted their "Radiax" slotted solid outer conductor cable to the AM band for highway radio. This was originally developed for VHF systems in metropolitan subways. Tests conducted under a Federal Highway contract several years along the approach road to Washington's Dulles Airport included Radiax, Halstead and our cable. The Radiax showed less down-line attenuation, as indeed it should in view of it's basic design for VHF and current price of $3.08 per foot. This cable might be a good choice if design considerations forced a particularly long cable. This happened in the case of a system recently installed along the Dulles Airport approach road where power for linear amplifiers was not readily available.

Leaky Cable

LPB and LocRad have a family of cables which resemble RG-58/U or RG-8/U with a very open outer braid. They are easy to handle and standard crimp-on connectors are available. Special types, such as the inclusion of a messenger cable for aerial suspension, or outer jackets of special color or composition, are readily avaialble. Cost of the larger cable, designated NF-2D, is about $1.00 per foot, depending upon length required.

Transmitter Systems

The transmitter is usually mounted well off the shoulder of the road in a weatherproof enclosure. At the end of a half-mile length of NF-2D the remaining power is down to nearly one-half that applied to the input end, as a result of combined cable losses and intentional radiation. Because of this, the radiated field adjacent to the far end of the cable is less than at the input. We speak of this as "pattern tilt," and find that a half-mile of NF-2D is about the practical limit in consideration of desired pattern tilt.

Licensed TIS Radiating Cable Systems

The above comments apply nearly 100% to licensed TIS radiating cable systems. The transmitter used must be FCC Type Approved for TIS service. The LPB TX2-30TIS was one of the first to receive this approval after the TIS Rules were adopted in 1978. It is used in all of the TIS systems we describe here, either radiating cable or vertical antenna.

Part 90.242 restricts a cable system length to 3.0km, or 1.9 miles. Co-channel stations operating under different licenses must be separated 0.31 miles. This is only 1,637 feet, which says something about the "packing density" available with cable systems in metropolitan areas.

Licensed Vertical Antenna Systems

The Rules limit the antenna height above ground to 49.2 feet. The resulting field strength is limited to 2mV/m at 0.93 miles, and the transmitter TPO is limited to 10 watts.

Reaching the permitted 2mV/m with only 10 watts requires some ingenuity in antenna design and attention to the ground plane, especially at 530kHz. For example, we have a center-loaded vertical whip which, with a 12-radial ground plane extending 100 feet, will typically produce 1mV/m at a mile with 10 watts input. It mounts on a 30 foot utility pole. For comparrison, a 44 foot self-supporting spiral-wound tapered fiberglass antenna at a far higher price will typically produce 4mV/m with 10 watts input.

A companion 1610kHz center-loaded vertical whip, with a similar ground plane, will meet the 2mV/m limit with 10 watts input.

Again, the transmitter is located at or near the antenna, and an antenna tuning unit must be located in a weatherproof enclosure at the base of the antenna.

AUDIO FOR HIGHWAY RADIO

The audio portion often seems to be the orphan child of a highway radio system. Unlike commercial broadcast facilities who are trying to hold their audiences for long periods of time, the highway radio listener, or potential listener, is usually in the capture zone for a very brief time. One or a series of very short messages that repeat continuously are required. Carts, continuous loop cassettes and more recently digital storage devices, are typically used for this audio. Each message source has it's advantages and it's own set of problems.

With budgetary constraints often paramount, cassettes appear to be an attrractive option. A good quality direct drive machine costs about the same as a single cart deck playback, and serves both record and playback functions. However, continuous loop cassette tapes in constant use don't last very long, and the message must be constructed to time to the length of cassette to avoid dead air. Heads also need to be cleared frequently.

Cart machines are the most common audio source in highway information radio systems, but digital storage units are gaining in popularity as this area of technology advances. By using double or triple deck playbacks, or single decks in tandem, one deck recues while the other one plays, so the need for precise timing of the message to avoid dead air is removed. Many facilities try to economize by using just one R/P unit, timing the message to fit the cart and shutting down when new carts need to be recorded. We are not always successful in discouraging this approach. Poor maintenance also often leads to customer complaints. It is somehow very hard to convince the user that he has to clean the heads and pinch rollers frequently!

Digital storage devices do overcome the maintenance and tape breakdown problems, but many produce less than broadcast quality sound. Most have been designed for telephone use, and therefore have been designed for limited bandwidth ... 300 to 3000 Hz. With the 6A3 emission requirement of TIS systems, the combined rolloffs can have a significant negative impact on sound quality. Bandwidth can be increased by adding memory or by reducing the message length capacity. Manufacturers differ in their expansion capabilities. There are units that can provide up to six minutes of messaage with bandwidth selectable up to 100 - 5000 Hz, but they carry a high price tag. The degree of user frendliness is also a factor in the selection of digital devices, and there is a lot of variance in this area.

The transmitter and the message origination center can be far enough apart to require phone line linkage. The choice of dial-up vs: broadcast quality lines, unfortunately, is often dictated by the price tag.

TIS was created as a service to the public. Poor sound quality and dead air reduces it's listenership and truncates effectiveness.

TIS FOR BROADCAST CONVENTIONS

Prior to the April 1986 National Association of Broadcasters Convention in Dalls, NAB inquired of the possibility of our supply of a temporary TIS system to expedite parking in the Dallas Convention Center area. The Dallas-Fort Worth Airport uses a TIS system on 530kHz for which they apparently have a power waiver allowing the use of one of our 50-watt transmitters. This made the competitive on-frequency in the Dallas area difficult, but our temporary system located atop the Convention Center provided an effective range of about 1.5 miles. The system was not as well advertised as we would have liked, but we trust it was of value.

A similar TIS system was temporarily authorized and demonstrated at the 1986 SBE National Convention in St. Louis, MO.

Again, prior to this 1987 NAB Convention, NAB requested a repeat of last year's TIS to assist Convention parking. It is now operating, and interested persons are invited to tour the elaborate studio facilities and the antenna/transmitter package on the roof of the Convention Center following this presentation. If you wish to discuss any aspects of highway information radio, please meet with us at any time during the Convention at the LPB Inc. exhibit, booth 3338, on the main exhibit floor.

We hope that this presentataion, plus your ability to listen to the TIS system operating here, will provide you an introduction to this broadcast-related specialty field.

THE NAB FM TRANSMISSION SUBCOMMITTEE — A REPORT ON ITS FORMATION AND TASKS

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The newly formed FM Transmission Committee has been commissioned, by the NAB, to study present and upcoming technical trends affecting FM broadcasting.

This paper will highlight some of the activities planned by this committee as it attempts to chart a course toward FM's future.

Commercial FM broadcasting began in the 1940's. After two decades of very slow growth, this new broadcast medium began to mushroom in the 1960's.

Experimental programming, relatively free of commercials and the introduction of stereo broadcasts attracted listeners by the thousands.

As FM receiver technology continued to improve, more and more licenses were granted. The listener shift away from AM quickly made FM the broadcaster's mainstay.

We broadcasters now face a situation similar to that of AM some 50 years ago. Everyone wants a piece of the action, but there are only a limited number of allocations.

A choice needs to be made. Are we going to limit FM station growth in saturated FM markets? Or, are we going to relax interference standards and allow more opportunities for new FM broadcasters?

This is the basis for one project currently being considered for study by the FM Transmission Committee. Interference is something that must be dealt with very carefully. The future of FM depends on accurately predicting the effects of interference upon our listeners. How much can a person tolerate before he switches to cassette, CD, or AM?

Listening to the standard AM broadcast band above 1200 kHz vividly demonstrates the interference problem at its worst. If we do not establish a strong framework for allocations, the same troubles may someday plague FM.

Indeed, a quick check of

the "educational" portion of the FM band in many parts of the country shows almost complete saturation. Stations are fighting over this 4 mHz piece of spectrum with a variety of operating powers and directional antennas.

If we look back at how the interference problem began on the AM band, it is not hard to imagine the same thing happening on FM. Unchecked allocations of new FM stations could certainly threaten the future of all FM broadcasters.

Not only should we learn from mistakes made during the early days of AM but also, we can take lessons from what is happening with TV.

High definition television is around the corner. In its present form, additional spectrum is needed to transmit all the video information necessary to construct the high definition picture. Broadcasters are asking for solutions to this problem.

An equivalent FM scenario may someday find us with the technology to vastly improve the quality of FM transmissions. But, this technology may be useless with a spectrum that is overpopulated and loaded with interference.

Most of the topics we are looking at as we study FM will either concern improving the technical quality of the service or increasing the number of stations. The goal is to prevent the AM'ization of our FM service.

FMX is a concept that has the potential to dramatically improve the service areas of many FM stations. Theoretically, a stereo stations coverage could almost extend to its equivalent mono coverage contour without the severe noise penalty now paid by stereo FM broadcasters.

FMX is the topic of another paper to be delivered later at this meeting. The FM Transmission Committee hopes to set the course for future development of this new technology.

Several proposals have been submitted to the FCC with regard to improving the competitive position of Class A stations against that of more powerful Class B or Class C facilities.

The committee will be looking at the pros and cons of these proposals as we weigh them against the future of the service.

One proposal, by Clear Channel Communications, asks for an across the board increase in power for Class A stations. Others, request increases in facilities only for those Class A stations with room to expand their coverage area without causing increased short spacing to other facilities.

There is also a proposal to create an entirely new FM service, called the FM2 band. The FM2 band would be in the 200 mHz range. According to its proponents, this new piece of FM spectrum would allow daytime AM stations the ability to better serve their listeners.

The committee will also look at FM broadcast antenna standards. Some groups are promoting the use of directional antennas to allow greater interference protection or more stations.

This is a very complicated issue. One which may require the assistance of educational broadcasters, who have been experimenting with directional antennas for some time. Again, our aim is the future well-being of the FM broadcast service.

The work contemplated by this committee is formidable. We envision having to commission research studies and work by groups outside of our committee to meet our commitments.

There was once a time, not too long ago, when in many sections of the country tuning across the FM band would only yield three or four listenable stations. Those days are gone. We are witnessing the rapid maturity of FM.

I'm sure if Major Armstrong were standing here today he would be urging broadcasters to work with the FCC, as a team, to ensure the long range viability of the medium.

This is the ultimate goal of the FM Transmission Committee and we encourage your comments on items that you feel concern the future of our industry.

MATCHING FM ANTENNA PATTERNS TO THE DESIRED COVERAGE

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ABSTRACT

Broadcast coverage is determined in great part by the pattern of the transmitting antenna, but antenna patterns are often somewhat confusing to engineers not accustomed to them. This paper provides a basic explanation of what antenna patterns are, what they show, and how they are related to the coverage achieved by the broadcast station.

INTRODUCTION

The coverage achieved by a broadcast station is in great part determined by the pattern of the transmitting antenna. This fact makes a careful consideration of antenna requirements imperative when contemplating a new station or antenna upgrade; however, many engineers not accustomed to dealing with antennas on a day-to-day basis find antenna patterns and terminology somewhat confusing. This paper is intended to provide these people with an introduction to antennas and antenna pattern basics and to show the relationships between antenna patterns and the resulting station coverage.

ANTENNAS AND RADIATION - SOME BASIC THEORY

The primary functions of an antenna are to efficiently radiate power from a transmitter into the space around its antenna and to concentrate the radiated power in desired directions so that it is not wasted in directions where there is no audience. The second function, that of concentrating the radiated power in desired directions, is the one with which we are concerned.

In order to visualize how an antenna concentrates power in certain directions, let's start at a rather illogical place: let's start with a theoretical antenna that radiates equally in all directions, called the isotrope or isotropic radiator. Imagine an isotrope sitting in the middle of the air somewhere and radiating P watts of power. Since the isotrope radiates equally in all directions, the P watts of power are spread evenly in all directions, a lot like light leaving the sun. Now, let's stretch the imagination even further by surrounding the antenna with a huge ball or sphere made of some radio-opaque absorbing material and having a radius of R meters; put the isotropic antenna right in the middle so that the antenna is R meters from the sphere's inner surface in all directions. The radiated power cannot penetrate the surface and, therefore, is stopped and dissipated on that surface by creating currents (I) over the surface which then dissipate the power as I^2R losses in the surface resistance of the sphere. Since all P watts were radiated and nothing between the antenna and the sphere absorbed power, P watts are dissipated over the sphere's surface. The surface area of the sphere is $4\pi R^2$ square meters, and there is P watts spread out (and dissipated) over the $4\pi R^2$ square meters, so we can define a power density, S, as watts per square meter

$$S = \frac{P}{4\pi R^2} \quad \frac{\text{Watts}}{m^2}$$

If we were to take away the absorbing ball, what would change? There would no longer be anything to dissipate or stop the radiating power, but the power would still be there. If we went out to a point R meters away from the isotropic antenna and measured the power density, we would still find $P/4\pi R^2$ watts per square meter. If we went to a distance of twice R and took a measurement, we would find that the power density would be one fourth as large as it was at R.

What we have just demonstrated is one of the fundamental laws of radiation: power density (often just stated as power) falls off as the square of distance from the source. This is true for any radiation whenever the point at which we are observing the radiation is far from the source of the radiation. There are some other interesting relationships that are true for radiation far from its source, but first, we have to define just what "far from the source" means.

As most of us know, associated with any frequency is a wavelength, λ, found by

$$\lambda = \frac{C}{f}$$

where C is the speed of light in whatever units wavelengths are desired, and f is frequency in Hertz (cycles per second). Associated with any antenna we can find the maximum physical dimension, D, (length, width, height) in the same units as λ. Finally, we define the distance from the antenna to the point at which we wish to observe the field as the radius of observation, r. Whenever r is greater than a particular distance (to be shown below) a number of approximations and simplifications can be made in the otherwise complex fields of an antenna. The distance beyond which these simplifications are true is referred to as the far field region of an antenna and includes all distances, r, for which

$$r \geq \frac{2D^2}{\lambda}$$

where D and λ were explained above. As an example of calculating this for an FM antenna, let's see where the far field region begins for a five-bay antenna operating at 98 MHz. The bay spacing is approximately one wavelength which, at 98 MHz, is 10 feet. This makes a five-bay antenna about 40 feet high and gives us the largest linear dimension, D = 40 feet. With D and the wavelength known, we can find the beginning of the far field region

$$r = \frac{2D^2}{\lambda} = \frac{2(40)^2}{10} = 320 \text{ feet},$$

Beyond 320 feet, we can use any far field relationships with confidence that they will be reasonably accurate.

We previously derived one far field relationship, the inverse square law relating power density (S), total radiated power (P), and distance (r)

$$S = \frac{P}{4\pi r^2}.$$

Let's look at a couple of others. Most of us remember Ohm's law for circuits as if it was permanently etched on the inside of our eyelids, but here it is again, just to refresh everyone's memory:

$$V = IR, \quad \text{or } I = V/R, \quad \text{or } R = V/I$$

where V is voltage across some resistance, R, and I is the current through the resistance. Some of you may be asking yourselves what this has to do with antennas. Simple: in the far field region of an antenna, the electric field (E), magnetic field (H), and free space impedance (Z_o = 377 ohms) follow the same relationship as the circuit parameters so that, at every point in an antenna's far field,

$$|E| = |H|\,Z_o \text{ or } |H| = |E|/Z_o, \quad Z_o = |E|/|H|$$

where the double bars, $|\,|$, indicate magnitude of the enclosed quantity. Also analogous to circuit theory is the relationship between power density (S), electric field (E), and magnetic field (H). In circuit theory, voltage, current, resistance and power are related by

$$P = V\,I = V^2/R = I^2 R$$

and, in electromagnetics, S, E, H and Z_o are related by

$$|S| = |E|\,|H| = |E|^2/Z_o = |H|^2 Z_o$$

From this last equation and the first one we derived (using S, P and r), we can find

$$|S| = \frac{P}{4r^2\pi} = \frac{|E|^2}{Z_o} = |H|^2 Z_o$$

$$|E| = \sqrt{\frac{Z_o P}{4\pi r^2}} = \frac{1}{r}\sqrt{\frac{P\,Z_o}{4\pi}}$$

$$|H| = \sqrt{\frac{P}{4r^2\pi Z_o}} = \frac{1}{r}\sqrt{\frac{P}{4Z_o\pi}}$$

These equations show two things: first that the power density at any point is proportional to the square of the E or H fields, and that the E and H fields are proportional to the square root of power radiated; and second, that the E and H fields decrease as a function of 1/r.

Now that we have covered some of the basics of radiation, we can move on to see how these relate to antenna patterns and coverage.

ANTENNA PATTERNS

We previously used a theoretical antenna, the isotropic radiator, in some of our explanations. This antenna radiated equally well in all directions so that a pattern would really show very little information. No buildable antenna, however, has an isotropic pattern, nor is one desired as a rule. So, for real antennas, we need to be able to describe graphically the radiation pattern in order to determine how the antenna concentrates the radiated power.

We have a problem to solve when trying to show an antenna's radiation on paper: a piece of paper is suited for displaying two-dimensional information, but since an antenna radiates in three dimensions, we need some way of showing three dimensions using only two. The answer, of course, is to use two paper plots to represent each antenna pattern. Although other plots are possible, the two most common are the elevation and azimuth patterns. If we could see RF energy like we can see light, the elevation pattern would be a "picture" of the shape of radiation intensity from an antenna taken from beside the antenna and looking at it horizontally; the azimuth pattern would be a "picture" of the shape of radiation intensity as seen from far above the antenna looking down. Let's take a few minutes to look a bit into how antenna patterns are measured.

ANTENNA PATTERN MEASUREMENTS

We have established that an antenna radiates power into the air, or free space, in the form of electromagnetic waves consisting of electric and magnetic fields. Most of us depend for our livelihoods upon the fact that a second antenna will feel some effect from these fields allowing some of the radiated power to be received. In fact, we can receive this radiated power and actually calculate the power density (in watts per square meter) or field strength. So, we can just walk around an antenna with our antenna and field strength meter, writing down the readings, and later, in the comfort of our offices somewhere, plot the antenna pattern. Right?

Almost right. We are reading volts per meter, or amps per meter, or watts per square meter from our instrument. From this, we can make up an azimuth pattern that shows the variation of field strength (or power density) with angle around the antenna . . . at some distance, r, from the antenna . . . and with some power, P, being radiated from the antenna. What happens if we would like to compare two different antennas? We either must make some rather complicated corrections to compensate for measurements taken at different distances and powers, or we must ensure that all conditions (other than the antennas themselves) are identical when measurements are made on the two antennas; or, we normalize both patterns.

Recall that the power density, S, had the form

$$S = \frac{P}{4\pi r^2}$$

for the isotropic radiator. This was true because the isotropic radiator radiated equally in all directions. Now, we are talking about a non-isotropic antenna that radiates more in some directions than in others. This means that S depends not only on P and r, but on azimuth and elevation angles, ϕ and θ, respectively. We can indicate this by writing S at ϕ_0 degrees from true North and θ_0 degrees above horizontal as

$$S \text{ at } \phi_0 \text{ and } \theta_0 = S(\phi_0, \theta_0)$$

and, from this, find a definition of angular power gain, G, for a 100% efficient antenna

$$G = \frac{S(\phi,\theta)}{S_i}$$

where S_i is the power density of radiation from an isotropic antenna fed with the same power as the antenna being measured and measured at the same distance, r. In other words,

$$S_i = \frac{P}{4r^2\pi}$$

and

$$G(\phi,\theta) = \frac{S(\phi,\theta)}{P/4r^2\pi} = \frac{S(\phi,\theta)4r^2\pi}{P}$$

Notice that, for distance (r) and input power (P) constant, the gain is simply a constant ($4r\pi^2$/P) times the power density, $S(\phi, \theta)$. Therefore, if we can measure $S(\phi,\theta)$, we can find the gain. This is what an antenna pattern range does.

A receiving antenna can be classified according to its "effective aperture," which is related to its gain. If a particular antenna has an effective aperture of D square meters and is held in a field having a power density of S watts per square meter, the antenna will produce an available power at its terminals of

$$P = SD \text{ watts}$$

which can be fed to a receiver. If we have a receiver from which we get an output signal proportional to the input power, we can write

$$X_{out} = C_1 P_{in} = C_1 DS = C_2 S$$

i.e., the output signal is simply some constant times the power density. If the receiver output is proportional to the input voltage or current (as is the more usual case), we can write

$$X_{out} = C_3\sqrt{P_{in}} = C_3\sqrt{DS} = C_4\sqrt{S} = C_5 E$$

where E is the electric field in the area of the antenna corresponding to a power density of S ($E = \sqrt{SZ_0}$) in volts per meter and C_5 is a constant that contains all conversion factors between E and X_{out}. In either case of receiver output, we have a value for S (either $X_{out} = C_2 S$ or $X_{out}^2 = C_5^2 Z_0 S$). If we take an antenna and spin it horizontally (i.e., varying the angle ϕ), the variation of the receiver output indicates the antenna azimuth pattern; if we spin the antenna vertically, the receiver output indicates the elevation pattern. All we need do is plot the receiver output versus angle, which brings us to our main point, normalization.

If we were to simply plot the receiver output, X_{out}, we would have a plot of output (in volts, amps, watts or whatever) versus angle. But remember, X_{out} is proportional to either $S(\phi,\theta)$ or $\sqrt{S(\phi,\theta)}$, and since $S(\phi,\theta)$ is proportional to input power and inversely proportional to the square of the distance, so is X_{out}, and we have not succeeded in our original goal of producing patterns (of different antennas taken on different ranges) that are easily compared. However, at some point or points, the power density, $S(\phi,\theta)$, will be maximum. Call this power density S_{max}. If we divide the power density at all points in the pattern by the maximum power density, S_{max}, the result will always be between 0 and 1. So, define $A(\phi,\theta)$ as the normalized antenna pattern defined by

$$A(\phi,\theta) = \frac{S(\phi,\theta)}{S_{max}} = \frac{G(\phi,\theta)\ P/4r^2\pi}{G_{max}P/4r^2\pi} = \frac{G(\phi,\theta)}{G_{max}}$$

and $0 >/= A(\phi,\theta) </= 1$. Notice that, not only is $A(\phi,\theta)$ normalized such that it is always between zero and one, but it does not depend at all on the range at which the pattern was measured or on the power used to take the pattern. This normalization can be performed using either power density or the electric or magnetic field. This can be shown by using the relationship

$$S(\phi,\theta) = \frac{|E(\phi,\theta)|^2}{Z_o}$$

in our expression for normalized pattern, $A(\phi,\theta)$.

$$A(\phi,\theta) = \frac{S(\phi,\theta)}{S_{max}} = \frac{|E(\phi,\theta)|^2/Z_o}{|E_{max}|^2/Z_o}$$

$$= \frac{|E(\phi,\theta)|^2}{|E_{max}|^2} = \frac{G(\phi,\theta)}{G_{max}}$$

Define $a(\phi,\theta)$ as the normalized electric field pattern,

$$a(\phi,\theta) = \frac{E(\phi,\theta)}{E_{max}}$$

and we find that

$$A(\phi,\theta) = \frac{S(\phi,\theta)}{S_{max}} = \frac{|E(\phi,\theta)|^2}{|E_{max}|^2} =$$

$$[a(\phi,\theta)]^2 = \frac{G(\phi,\theta)}{G_{max}}$$

or $$a(\phi,\theta) = \sqrt{A(\phi,\theta)} = \sqrt{\frac{G(\phi,\theta)}{G_{max}}}$$

This provides us with the relationship between power patterns and field patterns.

Normalized patterns of any number of antennas from any number of ranges can now be easily compared - as long as they are plotted identically! There are three common scales used to plot antenna patterns: linear field, linear power, and logarithmic or dB. Linear plots are typically plots of $a(\phi,\theta)$ or $A(\phi,\theta)$ on circular (for azimuth) or semicircular (elevation) paper marked off in angle (either ϕ or θ) and radial distances from 0 at the center to 1 at the outer edge. The levels are often read as percent field or percent power (i.e., a level of .5 is equivalent to 50%). The relationship between linear field, $a(\phi,\theta)$ and linear power, $A(\phi,\theta)$ is given by the equations

$$a(\phi,\theta) = \sqrt{A(\phi,\theta)}$$

and

$$A(\phi,\theta) = [a(\phi,\theta)]^2.$$

What this means is that, if a field plot shows a value of $a(\phi_o,\theta_o) = .4$ at some angles ϕ_o and θ_o , then the relative power at this point is

$$A(\phi_o,\theta_o) = [a(\phi_o,\theta_o)]^2 = [.4]^2 = .16.$$

Going in another direction, if $A(\phi_o,\theta_o) = .5$, then the relative field value can be found by

$$a(\phi_o,\theta_o) = \sqrt{A(\phi_o,\theta_o)} = \sqrt{.5} = .707.$$

The third type of plot is the logarithmic or dB plot; call the dB plot $B(\phi,\theta)$. At each angle of ϕ and θ , $B(\phi,\theta)$ is simply ten times the log of the normalized power pattern, $A(\phi,\theta)$. Because of the square relationship between power, $A(\phi,\theta)$ and field, $a(\phi,\theta)$, plots the dB value, $B(\phi,\theta)$ can be found from $a(\phi,\theta)$ by finding ten times the log of $a(\phi,\theta)$ squared or twenty times the log of $a(\phi,\theta)$,

$$B(\phi,\theta) = 10\log[A(\phi,\theta)] =$$

$$10\log[[a(\phi,\theta)]^2] = 20\log[a(\phi,\theta)].$$

Figures 1, 2, and 3 show examples of the same pattern plotted in field, power and dB just to show how dramatic the changes in appearance can be between the three types of plots.

The three types of patterns are used in much the same way. Let's say that we have a rating of 100 kW E.R.P. The E.R.P. is actually an abbreviation for Effective (isotropically) Radiated Power. The 100 kW ERP indicates that, at the angle of maximum gain, at any radial distance from the antenna, there should be a power density equal to the power density that would exist

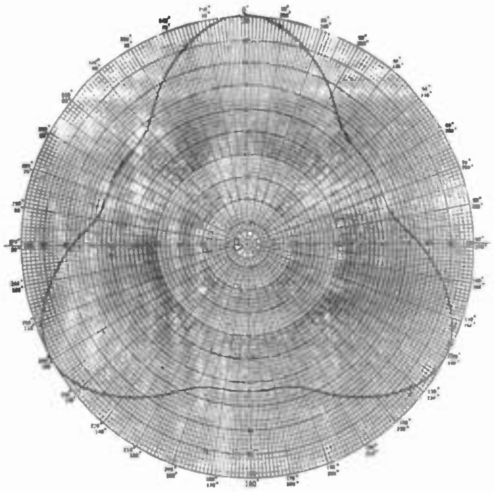

FIGURE 1

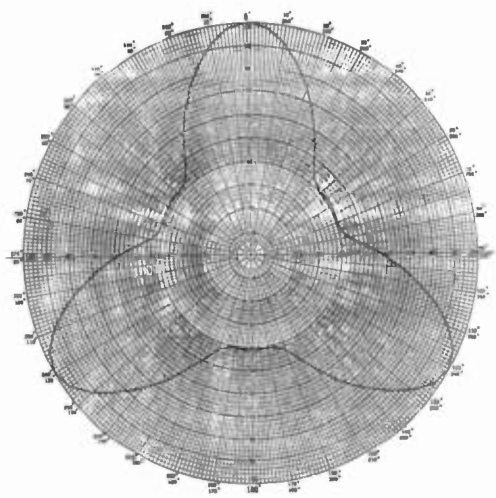

FIGURE 2

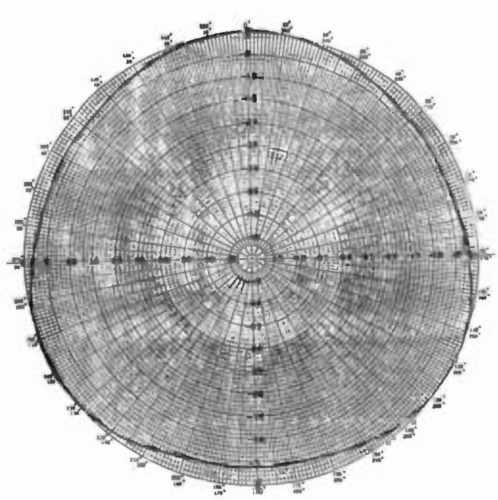

FIGURE 3

at that point if 100 kW were radiated from a theoretical isotropic radiator in free space. In other words, with 100 kW ERP, we should be able to go anywhere in the main beam (where the normalized pattern is 1) at any distance, r, and measure a power density, S, of

$$S_{max} = \frac{100\ \text{kW}}{4r^2\pi}$$

Using our relationships between S, E, H and Z_o, we can find equivalent electric and magnetic field strengths:

$$E_{max} = \sqrt{SZ_o} = \sqrt{\frac{(377)(100\ \text{kW})}{4r^2\pi}}$$

$$H_{max} = \sqrt{\frac{S}{Z_o}} = \sqrt{\frac{100\ \text{kW}}{(377)4r^2\pi}}$$

Once we have found S and E, we can then use the linear $A(\phi,\theta)$ or $a(\phi,\theta)$ plots to find $S(\phi,\theta)$ or $E(\phi,\theta)$ at any other angles but at the same distance simply by using

$$S(\phi,\theta) = A(\phi,\theta)\ S_{max}$$

and

$$E(\phi,\theta) = a(\phi,\theta)\ E_{max}.$$

The dB plot is used in much the same way. As an example, let's say that our antenna pattern, $A(\phi,\theta)$, is one at some point (some ϕ_o, θ_o) corresponding to a direction towards our city of license, which is 30 kilometers away. At this point (30 km or 30,000 meters) from the antenna, we find S_{max} to be

$$S_{max} = \frac{100{,}000}{4\ (30{,}000)^2} = 8.8 \times 10^{-6}\ \frac{W}{m^2} =$$

$$8.8\ \frac{uW}{m^2}$$

So, in our city of license, at this point, we should have about 8.8 microwatts per square meter. Using

$$E_{max} = \sqrt{Z_o\ S_{max}}$$

we find that this corresponds to about 58 millivolts per meter of electric field. We can then use this to find our signal strength at the same point in dBu's, which are defined as twenty times the log of the electric field strength divided by one microvolt per meter

$$E_{max}\ \text{in dBu} = 20\log\ \frac{E}{1\text{x}10^{-6}} =$$

$$20\log\ \frac{58 \times 10^{-3}}{1 \times 10^{-6}} = 95\ \text{dBu}.$$

Now, suppose that there is a second city located 30 km away from the antenna, but 65° away from the first city in azimuth. We look at our antenna pattern and find $A(\phi,\theta)$ for this direction to be $A(\phi_o,\theta_o) = 0.43$. This corresponds to

$$a(\phi_o,\theta_o) = \sqrt{A(\phi_o,\theta_o)} = \sqrt{.43} = .66$$

$$B(\phi_o,\theta_o) = 10\log[A(\phi_o,\theta_o)] = 10\log[.43] =$$

$$= -3.67 \text{ dB}.$$

Our ERP in this direction is simply

$$ERP(\phi_o,\theta_o) = A(\phi_o,\theta_o)[\text{rated ERP}] =$$

$$.43[100 \text{ kW}) = 43 \text{ kW},$$

our power density is found from

$$S(\phi_o,\theta_o) = S_{max}\, A(\phi_o,\theta_o) =$$

$$8.8 \frac{\text{u watts}}{m^2} [.43] = 3.78 \frac{\text{u watts}}{m^2},$$

our electric field strength is

$$E(\phi_o,\theta_o) = E_{max}\, a(\phi_o,\theta_o) = 58 \frac{mV}{m} [.66]$$

$$= 38.28 \frac{mV}{m},$$

and our field, in dBu, is

$$E_{dBu} = E_{max,dBu} + B(\phi_o,\theta_o) = 95 - 3.67$$

$$= 91.33 \text{ dBu}.$$

We have now seen how to take and use the antenna pattern to estimate what the approximate coverage will be. Remember, however, that raw antenna patterns, such as are normally received from the manufacturer, do not take into account attenuation or scattering due to the ground or surrounding terrain. The pattern measurement performed by an antenna manufacturer is a reasonable picture of an antenna's radiation if the antenna and tower were suspended in free space, far from the earth or any other obstacles. However, anything in the path of an antenna's radiation may perturb or change the pattern somewhat. Let's take a quick look at the general methods of interaction, and then go a bit deeper into a few specific radiation-obstacle situations.

SCATTERING AND REFLECTION

Just about everyone has, at one time or another, thrown a small pebble into a quiet pond or puddle and watched the concentric rings of ripples move away from the initial point of impact, slowly decreasing in amplitude as they move. This is a reasonably good two dimensional picture of radiation from the ideal isotropic radiator in free space. The radiation travels outward in all directions, diminishing in strength as it gets further from the source.

Now, think of the same pebble dropped into the water only a few inches from a wall. Initially, the ripples leave the point of impact just as they did when the wall was not nearby. But soon, the first ripples hit the wall. Depending on many factors (such as the wall shape, size and material), the ripples will partially or wholly reflect from the wall and begin moving back towards the source of the ripples, the place where the pebble landed. The backwards traveling ripples meet the forward traveling ripples and create interference patterns. Where two ripples -one forward, one backward traveling - meet with both ripples high (the ripple or wave level above the flat water or average water level), the resulting level is the sum of the two and is higher than either. Where two ripples meet with low (below average) water levels, the resulting level is lower than either of them.

Electromagnetic fields exhibit the same type of phenomena. As might be imagined, predicting the resultant pattern for a complex arrangement of antenna, tower and surrounding landscape is nearly impossible to any great accuracy. However, the antenna manufacturer can usually do a very good job of measuring the effects of the tower and its contents, as described below, and a consultant can utilize statistical curves, maps, computers and other tools to predict the effects of surrounding terrain. Now that everyone has a basic knowledge of what patterns are and how nearby objects can affect them, the rest of this paper will delve a bit deeper into some specific problems that commonly are seen.

ANTENNA PATTERN DISTORTION

Thus far, we have talked mainly about antennas in free space. In other words, antennas suspended in air without towers or other supporting structures of any kind. This discussion is of immense value when trying to understand and visualize how the antenna system produces the pattern required by broadcasters for use by their listening audiences. However, in order for these antennas to radiate signals to as large an area as possible, they must be physically positioned with respect to their intended listeners so that interfering structures such as hills and buildings will not absorb, reflect or otherwise block these signals from reaching the audience. Also, there must be a way of channeling the signal from the station's transmitter system to the antenna.

In order to overcome the adverse effects of buildings and mountains, antenna systems can be placed high above the area for which the broadcast is intended. This allows most of the transmitted signal to travel over these obstructions, unencumbered, to its destination. That is why antennas are usually mounted high on tall buildings or atop tall towers. The need for transmission of the signals from the station's transmitting plant to the antenna is filled by use of what is known as an antenna feedline. Signals exit the transmitter and are carried up the tower and distributed to the individual radiating elements that are physically attached to the feedline. We should now be able to visualize the entire configuration; the antenna elements mounted to the feedline, and that entire assembly bolted to a tower or other supporting structure. This is how most side-mounted FM antennas are constructed.

If we think for a moment about the radiation pattern from this type of system, we may begin to suspect that the feedline and tower, positioned so close to the antenna elements will have a profound effect. Electromagnetic radiation, which is exactly what a radio signal is composed of, is created or excited by time-varying or oscillating electric currents flowing in and around any physical structure that will conduct electricity. The station's transmitter produces a signal that causes oscillating currents to flow in and around the conducting elements of the antenna. These currents, in turn, produce electric and magnetic fields around the antenna that give rise to the radiated signal that travels out to the listeners. However, these fields around the antenna do other things too. Besides producing the intended radiated signals, the fields will induce currents in other structures that conduct electricity near the antenna such as the outside surface of the antenna feedline, so called ground straps along the antenna feedline that electrically bond the line to the tower, legs and support members in the tower and mounts that secure the antenna to the tower or other support structure. These induced secondary or parasitic currents will then produce electric and magnetic fields of their own. These secondary fields will interact with the primary fields from the antenna itself, producing a resultant field distribution that is altered significantly from the field distribution that would otherwise be present from the antenna alone. And, since it is this resultant field distribution that will determine the final antenna pattern, the need for a thorough evaluation of these effects should be apparent to insure that the resultant antenna pattern is suited to the needs of the broadcaster.

Figures 4, 5 and 6 are patterns measured on our range for three different side mount antennas. These patterns make evident the amount of interaction between the antenna and its immediate surroundings, and the necessity for evaluation.

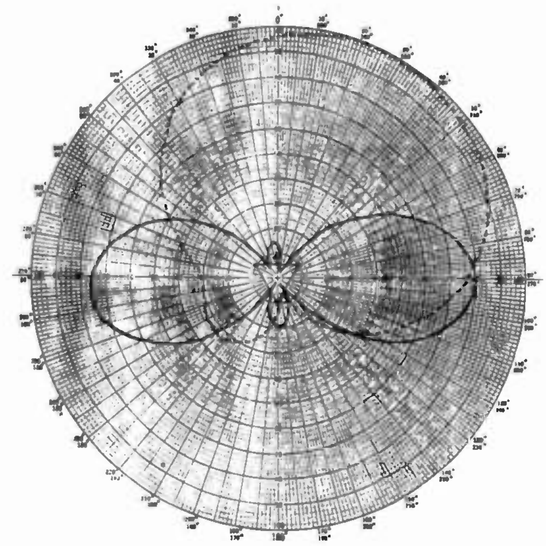
FIGURE 4

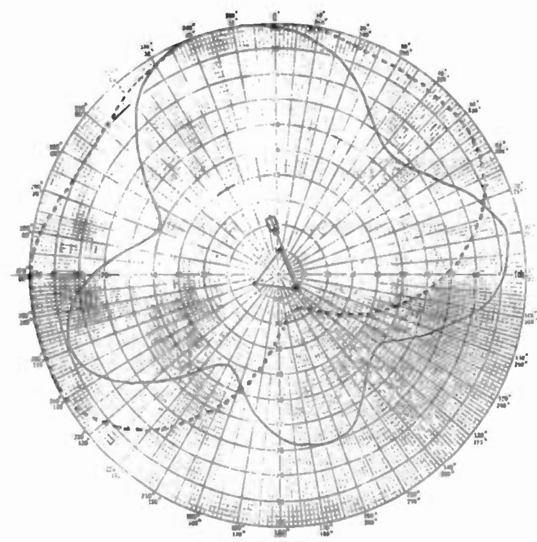
FIGURE 5

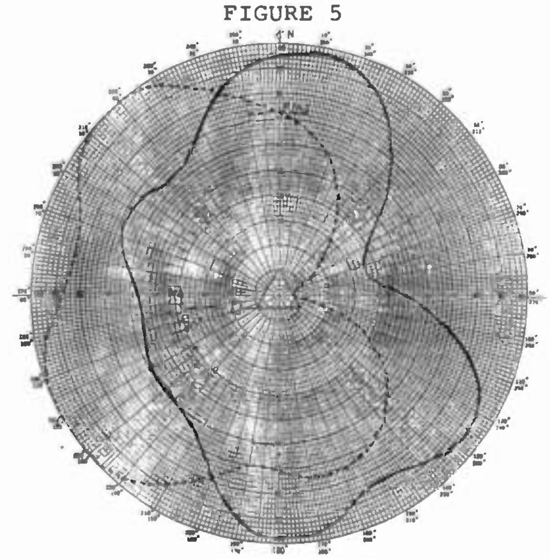
FIGURE 6

DIRECTIONAL ANTENNAS

In certain cases, this parasitic effect can be extremely useful. Antenna patterns can be manipulated greatly by the conducting elements near the antenna. These aptly named parasitic elements have no direct electrical connection to the antenna feedline. Their field-producing currents are induced by the fields from the feedline-driven elements in the antenna in exactly the same manner as the other conducting members do, as discussed earlier.

The type and degree of antenna pattern alteration by parasitic activity depends on several factors. Prediction of the pattern by analytical means from a given configuration may be done; however, it is quite complex and carries with it a certain degree of uncertainty. It requires the superposition of the complex fields from all current-carrying members in and around the antenna. These fields describe what are called radiation moments. Each radiation moment has an associated relative strength or magnitude, as well as an associated relative direction or phase. Because of the fact that these quantities have both a magnitude and a phase associated with them, they are said to be complex or vector quantities. Computer pattern calculation is done by summing all these radiation moments at each point in space around the antenna.

Directional antennas are implemented by introducing radiation moments from other antenna elements, usually parasitic, and allowing them to interact with the moments from the antenna elements driven by direct connection to the antenna feedline. The system is adjusted so that this interaction will take place in such a manner that the resultant antenna field distribution will produce a net reduction of the radiated signal in directions where too much signal is undesirable, such as government receiving installations, and an augmentation of the radiated signal in desired directions such as large cities where there are many listeners.

Figure 7 shows a representative pattern for an intentionally directionalized antenna.

One specific type of antenna that affords a high degree of control over the pattern is the so-called panel antenna. Panel antennas have incorporated within them a shielding panel or screen positioned between the antenna radiating elements and the tower, that greatly reduces the magnitude of parasitic currents in tower and support members. Panel antennas produce relatively distortion-free predictable azimuth pattern which are ideal in high quality installations or in situations where a specific directional pattern is mandated that cannot be achieved with a side mounted antenna.

Figure 8 shows the high level of omnidirectionality in azimuth which can be achieved with a panel antenna.

Directional antennas must be specifically applied for on the application for a construction permit with the Federal Communications Commission.

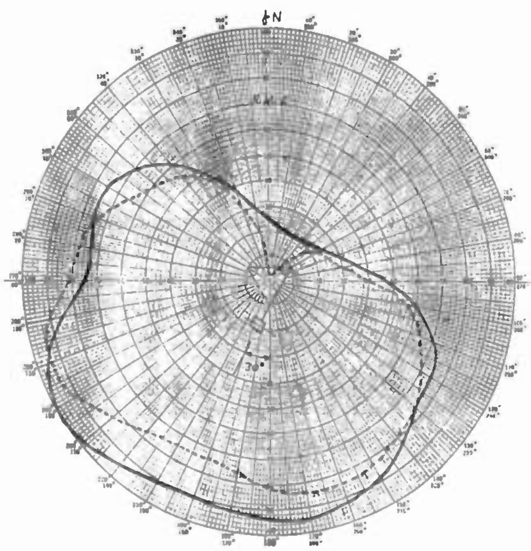

FIGURE 7

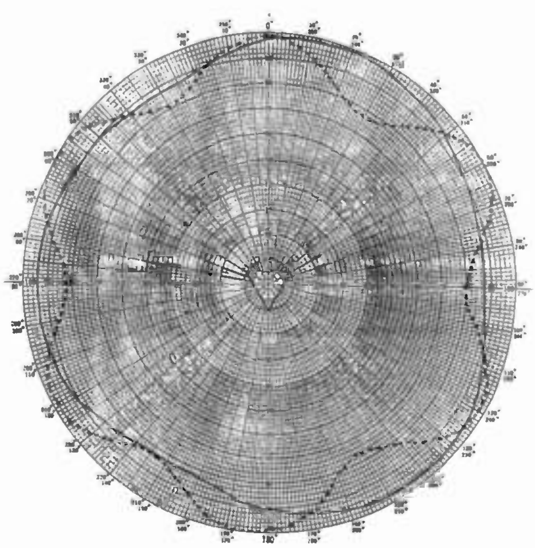

FIGURE 8

RANGE AND FIELD-MEASURED PATTERNS

Antenna patterns, though they may be numerically calculated as described above, are usually measured on an antenna pattern range. A single element of the antenna is mounted on a tower containing all conducting members and structures that may carry currents producing fields that can distort the pattern. The physical configuration of the entire antenna system, including feedline, feedline grounding straps, distance between the tower and antenna and parasitics, if any, is adjusted so that all radiation moments, when added together, produce the desired or required pattern. The pattern range itself must be constructed extremely carefully. As the signal is radiated and travels from the transmitting to the receiving antenna, the range must not introduce any parasitic effects of its own that will not be incorporated in the final antenna system in the field. Range parasitic effects may be suppressed by the careful placement of special radio-absorbing material around the test site so that an accurate pattern picture of the entire antenna system is assured.

Once the antenna is placed in the field, the pattern may be finally measured by actually flying around the antenna site at a constant distance with the proper recording instruments, and plotting the relative field strength as a function of azimuth angle. There should be little discrepancy between the pattern measured on the test range and that measured in the field if cautious measuring practices are carried out on a properly constructed pattern test range.

TERRAIN AND MULTIPATH

The overall performance of a broadcast station is influenced by other factors aside from the antenna system itself. The antenna may be producing a pattern that should provide a high level of signal over a given area, but when that signal is measured, the results show otherwise.

As mentioned earlier in this paper, large structures such as buildings and mountains will reflect and/or absorb signals from an antenna on their way to a listener's radio receiver. This was seen when we described waves on the surface of a pond and how they interacted when they encountered a piece of wood. If these obstructions are large enough, or if they are located in such a position with respect to the intended listening area, they absorb and/or reflect the signal, resulting in a reduction of this signal to unacceptable levels. In this situation, unless the antenna site is relocated, there is little that can be done about these obstructions outside of using a D-9 bulldozer or a freight train load of high explosives!

Another phenomenon that is a causal factor for ulcers is known as multipath. In some areas, mountains and buildings may set up a path for a signal to follow as it travels from the station's antenna to the listener's receiver. The signal leaves the transmit antenna, propagates until it hits a mountain or building, reflects and then moves on to the receiving antenna. At the same time, a signal that leaves the transmit antenna at a slightly different angle may have a clear line-of-sight path from the transmitter to the same receiver. It should be apparent that these two signals have taken two different paths on their way from the transmitter to the receiver. These signals each produce currents in the receiving antenna that are passed on to the receiver circuits. These currents are parasitic currents, and as such, are vector quantities, as discussed earlier. The magnitude and phase of these currents induced in the receiver antenna are determined, among other things, by the length of the path from the transmitter to the receiver. If the two paths taken by each signal are of lengths such that the receive antenna currents resulting from the each signal are nearly equal in magnitude but opposite in phase, the signal will cancel in the receive antenna and never reach the receiver circuits. The result is an apparent loss of signal.

In some cases, the antenna pattern may be adjusted to reduce the effects of multipath, however, in the majority of severe cases, multipath must be dealt with using other options.

CONCLUSIONS

Whenever an antenna is built and mounted to a supporting structure, it is virtually impossible to exactly predict the resulting pattern characteristics without complex involved computer calculated or range-measured patterns. In this paper, we have seen how an antenna works, and have investigated some "real world" effects on patterns, and how the radiated field or power density at any specific point around an antenna is influenced. We are able to measure the effects of these factors and, to a large degree, control the resulting interaction of all the radiation moments in a system producing a pattern that is both legal and desirable. Because the Federal Communications Commission requires all patterns to be as omnidirectional as possible, with the exception of permitted directional antennas, pattern measurement becomes almost required. In any case, antenna patterns are extremely important, and the proper installation of an antenna system should include a thorough evaluation of these effects to insure many years of optimum, trouble-free performance.

FUNDAMENTALS OF TV AND FM BROADCASTING ANTENNAS

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Over the course of their careers, most broadcasting engineers never really get a chance to study antennas to the degree they would like. The incidence of requiring a new antenna is low, therefore when a need does arise, engineers tend to rely on the testimonials of others to help them with their choice. This paper hopes to improve on that process with a review of fundamentals to help the engineer ask the right questions of his potential supplier.

The list of characteristics of an antenna that are critical is:

A. Horizontal Radiation Pattern
B. Vertical Radiation Pattern
C. Polarization
D. Matching Techniques
E. Power Handling
F. Materials

Horizontal Radiation Patterns

Coverage is determined by the HRP and difficulties occur when the mounting structure electrically influences the pattern. Patterns that radiate through towers or reflect from them are only a temporary solution to an already compromised approach to antenna array design. Towers are forever being modified with new lines and re-inforcements so pattern stability is lost with the first change.

It is hard to understand but licensing procedures don't require factory pattern testing on omni antennas. They do require it on directional antennas. Consequently the omni antenna buyer is totally dependent on the supplier's advertised patterns which do not simulate his tower. Even top-mount patterns can have structural limitations when the strength of the mount must take priority over the pattern.

The accepted definition of omni is ±2dB while directional can be anything from as little as a fraction of a dB to many dB. The directional protection is usually an absolute ERP at an azimuth while omni is absolute only with respect to RMS gain which is an average of all azimuths.

Polarization is also greatly influenced by structure. Factory testing is really the only insurance when there is a question of structural influence on the HRP.

The greater the number of driven elements in a bay the greater the flexibility you have in HRP shapes in order to meet a specific need.

Vertical Radiation Patterns

The most important feature of an antenna is its vertical radiation pattern. The VRP determines the RMS gain of the antenna. The greater the concentration of energy on the horizon the greater the gain of the antenna. Energy anywhere else but on the horizon is a loss of gain and an inefficiency. Energy distribution from the horizon 0° down to let us say 45° affects the strength of the signal on the audience from the horizon down to the property line of the antenna farm.

People close to the tower don't need very much signal while those far away need all you can deliver. The shape of the ideal distribution is a cosecant curve and it means when you ½ the distance you should reduce to ¼ of the power. In a cosecant vertical distribution everyone has the same field strength regardless of distance.

High gain antennas broadcasting from great heights require beam tilt and null fill because of the curvature of the earth and the long distance one can see from great heights.

Minimum Beam Tilt for Smooth Earth:

For 1000' Angle is .559° and Horizon is 38.82 Miles.
For 2000' Angle is .79° and Horizon is 54.86 Miles.
For 3000' Angle is .968° and Horizon is 67.22 Miles.
For 4000' Angle is 1.117° and Horizon is 77.57 Miles.
For 5000' Angle is 1.249° and Horizon is 86.74 Miles.
For 6000' Angle is 1.369° and Horizon is 95.08 Miles.
For 7000' Angle is 1.478° and Horizon is 102.66 Miles.
For 8000' Angel is 1.58° and Horizon is 109.74 Miles.
For 9000' Angle is 1.676° and Horizon is 116.42 Miles.

For 10000' Angle is 1.767° and Horizon is 122.74 Miles.

Beam tilt and null fill are a result of vertical phasing in most modern antennas but the quality of that technique varies from manufacturer to manufacturer. When you transmit from the top bay first and then the next & the next, etc, each slightly delayed, you develop a beam tilt. If the delay is not equal steps you develop null fill as well. The trick is to do that with the minimum of a loss in gain. To learn how much gain you lose you need to see a VRP without beam tilt and null fill.

Sequential centre fed antennas have a problem achieving optimum null fill because of the fixed propagation distance (one lambda) they require between radiators. This type of antenna uses unequal power to produce null fill and beam tilt by offset centre feed so that the top half is slightly ahead of the bottom half.

Another point about sequentially fed antennas is the cumulative phase change with frequency. A 1% frequency change is a 3.5° phase change on the first bay and it multiplies with the bay number in the sequency ie: 8 bays 25.3°, 16 bays 50.6° etc.

Parallel fed bays where the energy to be radiated reaches the radiator in the same wavelength is very much superior design because the 1% change is 1% of the differential phase and not cumulative as it is in the sequentially fed antennas. All broadband antenna are parallel fed because even a 20% frequency change is only a 20% phase change of the differential phase. ie. 20% of 40° is 8°.

It is apparent that end fed sequentially fed antennas would have a changing tilt problem across their bandwidth because of cumulative phase.

The vertical pattern has become even more important in recent times with regard to non-ionizing radiation limits of $1mw/cm^2$. End fire radiation downward is important for the safety of people in and around the base of the tower. However if the height is higher than a few hundred feet even strong end fire radiation is no hazard but it still is an inefficiency and that means loss of gain. One should always ask to see the vertical radiation pattern out to 90°.

Polarization

This is a subject that really needs public discussion. Not so much the linear horizontal polarized signal but rather the much touted so-called circular polarized signal.

A definition is -

CP is the resultant constant amplitude single rotating vector produced at all points in the coverage area where two equal, linearly polarized, orthonganally related signals, arrive with 90° phase difference.

It would be good if all advertised circularly polarized signals did that. The easiest way to judge an antenna for its CP is to look at it from all azimuths and note if the mechanical appearance changes in relationship to you so will the CP. For instance if Radiator #1 and Radiator #2 are one ahead of another, do they remain in that order when viewed from a different direction. If not the then circular polarization must change.

In spiralled second radiator techniques to produce CP we have the sequentially fed aperture problem with excessive phase change with frequency. In large antennas the CP changes considerably across the channel with this technique.

Perhaps the fact that double power is permitted in CP transmission is really the more important aspect rather than the advantages of CP.

In FM true CP is an advantage in the concrete canyons of a city because most building surfaces are polarization selective in their reflectivity. A true circularly polarized signal will test all surfaces once a cycle for selective reflectivity and generate multiple standing fields of signal in the streets to yield a more uniform net field than that obtainable with a linear signal. Short distance reflections less than 1 KM differential path length are not distortive because they only increase or diminish the main field. Multiple reflections are better than singular because the deep nulls are averaged out. On the other hand long distance reflections greater than 1 KM into shadowed areas can be disasterous. The same signal arriving that late will cause harmonic distortion in the receiver. A separate paper on FM Multipath will be published shortly. Its title is "Multipath in FM Broadcasting".

Matching Techniques

Television broadcasters are keenly aware of the signal degradation caused by reflections from transmission antennas. The effect is greater in longer feed systems because the delay is greater and it is easier to see.

In television the signal is time division multiplex, ie: when there is sync there is no vision and when there is vision there is no sync. TV signals therefore require a degree of freedom from reflection akin to the dynamic range of the vision sidebands.

Subjective tests reveal that coherent interference 54 dB below carrier is the limit of perceptability. It follows with sidebands 17dB below carrier we have a 37 dB dynamic range in the vision intelligence. Theoretically adequacy would be a match of 37 dB return loss or 1.0287:1 VSWR. The industry has adopted a 1.05:1 as a practical value because of the difficulty especially at higher frequencies to obtain feed systems with 37 dB return loss. However please note 1.05 should apply to the whole band and not just to the areas around the vision carriers to allow for weather effects. In FM matching is not so critical and 1.1:1 or better is adequate.

Fine matchers need to be discussed because many people expect miracles and they should not. In order for a fine matcher to work effectively, the antenna must exhibit a constant impedance over its bandwidth. This means that the antenna must have some compensation engineered into it so that the impedance does not wander very much over its range of frequencies. That impedance whatever it is then can be transformed to 50 ohms with a fine matcher.

If the antenna wanders in its inherent impedance with frequency then only one frequency in its range can be matched much like a resonant circuit. Sharply resonant antennas aside from their effects on multiplexed signals are problematical when weather effects are added.

This brings us to the final part of our discussion on matching - namely ice and snow effects. It may be obvious by now, a broadband antenna array is far less susceptible to environmental changes. However it goes even further because a broadband antenna can be designed to have absolutely no effects with ice and snow without the use of 3 dB couplers and their absorbing loads or de-icers or massive radomes.

The technique is called automatic VSWR compensation. It makes use of mechanical and electrical phase design to bring reflections back to junctions in the feed harness out of phase with one another. It can only be done if the radiating elements are normally well matched over a broad range of frequencies and the elements themselves are protected from ice and snow bridging drive points on the antenna.

Power Handling

It is always a good idea to review the connectors and cable sizes referring to manufacturer's ratings to assure yourself of the adequacy of design in an offer. Personally I like to see 40% or more safety factor because it is good insurance.

In main transmission lines where the safety factor is often too small one should consider twin feeders with slightly smaller size to regain the factor. Once again it is good insurance.

The cost of a replacement part is not the prime consideration but rather the cost of replacing that part. Tower riggers who live hard and play hard also retire early because they can afford it.

Materials

A broadcasting antenna should have a long life and withstand all the nasty things that mother nature throws at it. The elements have to be rugged and protected from corrosion. To date hot dipped galvanized steel is the best choice. Sometimes even that is overcoated to guard against salt laden air.

Harness has to be lashed down because the wind can damage loose cables. Cables should be tied to the underside of tower members to guard against falling ice and riggers boots.

DC grounded antennas are a big advantage because the current from lightning strikes travel the outer conductors to system ground rather than via inners which have trouble handling the currents.

Sumary

A Broadcasting Antenna is a Major Investment. To protect that investment - here are six suggestions:

1. Horizontal patterns should not electrically involve support structure for their shape,

2. Vertical patterns that are parallel fed are more stable with frequency and can be tailored to meet ideal null fill and beam tilt specifications,

3. Polarizations should be constant with azimuth and with frequency,

4. Matching that results from internal element compensation to make impedance constant with frequency is most desireable,

5. Power handling should be conservative,

6. Materials should be strong, corrosion resistant and well installed.

THE EFFECTS OF ATMOSPHERIC ANOMALIES ON VHF SIGNAL PROPAGATION

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ABSTRACT

The effects of weather on the propagation of radio and TV signals has been noticed since the beginning of broadcasting. Unfortunately and to the detriment of broadcasters and their audiences, these anomalies often go unidentified and their effects were not taken into consideration when the FM-TV broadcast station transmission tower sites are chosen. This has resulted in some stations in coastal areas and near large bodies of water experiencing considerable and seemingly unaccountable signal problems.

With proper understanding of the dominate geographic and weather features in a given area, a tower site can be selected that will afford a broadcaster the maximum possible resiliency to these signal anomalies.

INTRODUCTION

As the chief engineer for WUSA-FM, Tampa-St. Petersburg-Clearwater, Florida, I have often received "bad signal" reports from with-in our predicted 1mv primary coverage area. In talking with other area chief engineers I learned that all area stations have similar problems with their signals at certain times of the day and year. After talking to the local office of the National Weather Service it was found that the bad signal reports coincided with temperature inversions over the large inland bay in the center of our metro area. These inversions are usually a daily occurrence in the summer and winter months.

As part of the process of choosing the best possible site in which to relocate and upgrade the new WUSA-FM tower, it was necessary to identify and fully understand the phenomena behind these signal anomalies.

THE LABORATORY

The major geographical features of this area are the Gulf of Mexico and a large inland body of water, Tampa Bay. The many cities that comprise the Tampa Bay Metro Area are built around this Bay and over this bay nearly all broadcast signals must travel. The atmospheric disturbances caused by these features have long been identified as the cause of North America's most severe weather. As I have now documented, the presence of these bodies of water cause what could easily be America's most severe VHF-UHF propagation anomalies. This condition has given me an excellent "Guinea Pig" for FM and TV signal propagation research.

HELP FROM THE NATIONAL WEATHER SERVICE

Specific information about the area under study was needed, a call to the local National Weather Service provided some important information.

Clear nighttime skies produce radiation inversions where the surface looses heat faster than the air. This produces early morning temperature inversions lasting until about 9am, with variable height of 500 to 1000 feet.

The summer temperature inversions are usually found in the afternoon between 2pm and dusk. They are most common near the 1000 foot level. These are produced by the hot sunlight, cooler water and afternoon sea breezes.

A third type of inversion associated with intruding major weather systems is the frontal inversion, and will usually occur at between 3000 and 5000 feet.

Daily afternoon thunderstorms fed by an unlimited supply of moist tropical air also cause atmospheric disturbances.

METEOROLOGY

Some knowledge of meteorology is necessary for the understanding of atmospheric effects on signals.

Incoming solar energy (sunlight) arrives in the "short wave" mode. Once this energy hits the earths surface it is converted into "long wave" radiation (radiant heat).

Under normal conditions the atmosphere is heated from the point of contact with the land or water and then becomes gradually cooler as altitude increases. Under normal conditions the temperature and therefore the density and water content, make a gentle transition that does not have a major effect on FM and TV signals.

A temperature inversion is simply a reversal of the normal order of cool air above warm air. Inversions are formed when cool air intrudes under warm air. A very sharp boundary occurs with this phenomena because the cool dense air tries to descend and the hot air tries to rise, with neither side winning and a very sharp boundary in temperature and density being the result.

Bodies of water retain temperature longer than air or land and as such causes atmospheric density anomalies along the shorelines and inversions over it's surface. It is these inconsistencies that lead to problems.

In nearly all coastal areas an additional effect is found with-in 10 to 20 miles of the shoreline. On warm summer afternoons, moist cool surface level sea breezes blow in land and cause the previously heated air to rise above the cooler intrusive air, producing a temperature inversion.

The effects of media density on propagation can be demonstrated at light frequencies in many ways. You have probably noticed that a straw inserted into a half full glass of water seems to break in two as it passes through the surface of the water. This is because of the difference in media densities that the light passes through. Also notice that if you take a piece of clear glass and hold it up at a low angle to your line of site a partial mirror effect is produced, again due to the different densities of the media the light is encountering. You may also notice some channeling or tunneling where light picked up by the broad face or one edge of the layer of glass is redirected out the edges.

THE FCC 50/50 PROPAGATION CURVES

The FCC's FM & TV 50/50 predicted coverage charts were developed in the early days of FM & TV broadcasting and are the result of measurements over dry land.

In FCC Rules part 73.311, they warn you that the FCC 50/50 contours indicate only "the approximate extent of coverage over average terrain.....under actual conditions the true coverage may vary greatly from these estimates.......the estimated contours give no assurance of service to any specific percentage of receiver locations with-in the distances indicated".

Water based propagation has never been addressed by the FCC and apparently has never been tackled by broadcasters. It has erroneously been assumed that water provided the least signal damaging propagation because of it's low vertical profile. The atmospheric effects above water have previously been overlooked.

GOOD ENGINEERING PRACTICE

When designing any point to point radio frequency link it is prudent to take into account the maximum atmospherically induced signal fade and then engineer in enough extra signal level to overcome the depth of these fades. Achieving this fade margin is a relatively easy process involving antenna gain, worst case path loss, bandwidth, receive antenna gain, receiver sensitivity, and desired signal to noise ratio.

Point to point radio link designers will also usually allow for extra losses over water due to atmospheric conditions and the high reflectivity of water causing grazing path reflections (multi-path).

At broadcast frequencies all of this engineering is customarily thrown out the window and we revert to the FCC 50/50 charts to determine signal qualities at various distances. The tower is then located so that the theoretically predicted 1mv signal covers the city or market to be served.

Unfortunately, there are some significant signal losses possible that are not adequately addressed by the 50/50 charts.These losses are caused by atmospheric anomalies associated with large nearby bodies of water and are by no means limited to my area. However, their depth and frequency have made Tampa an excellent area for study.

DOCUMENTATION OF FIELD STRENGTH ANOMALIES

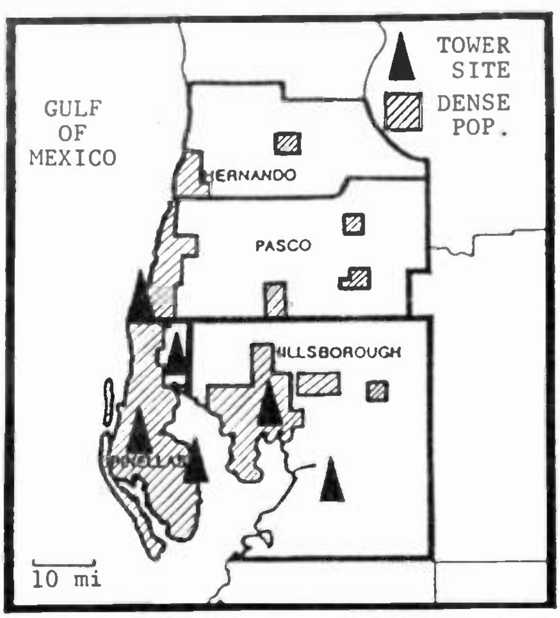
A systematic study of various broadcast tower to receiver path profiles were made in an effort to identify, understand, and document signal propagation.

Potomac Instruments model FIM-71 field intensity meters were used to continuously monitor received signal levels of FM carriers broadcast from specific height antennas over the various paths being evaluated. A Astro-med dual channel chart recorder operating at 30 mm per hour was employed as a recording device. The recorder was carefully calibrated to accurately display the .1 to 1 mV or 1 to 10 mv scales of the FIM-71s on the linear scale chart paper.

A fixed mounted vertical whip at a height of 20 feet was used as the receiving antenna, its signal was fed into a 3 db wideband splitter transformer with DC isolation (necessary for proper operation of the system). Each leg was directed through a 10 db pad to prevent any interaction of the input stages of the two field strength meters.

Because of the large geographic area of metro Tampa Bay , path lengths of 10 to 40 miles were involved. These distances allowed me to measure all broadcast signals over both land and mixed land-water routes. This is typical of the distances encountered between major cities and the most common broadcast tower locations in our area.

TAMPA BAY METRO



RESULTS

The studies show near predicted levels for land based propagation. Water based paths commonly show extreme departures with very unstable propagation, often 10 to 20 dB, below predicted FCC 50-50 figures. It was also found that tower height produced an inverted effect on coverage over water due to horizontal temperature inversion layers that reflect signals radiated from above the inversion boundary. Land to water transitions caused vertical temperature gradients that further affected signal levels.

An additional problem was noticed in the form of tropospheric ducting of signals along the sea coast from several hundred miles away. These strong imported signals would violate the theoretically protected 1mv primary coverage area contour.

Existing FM transmitter sites built on the bay's shoreline at times showed serious signal problems at and beyond the opposite shore, possibly due to multipath from the surface of the water and over water inversions and ducting. (Someone building a station in a similar location may do well to operate with a high gain antenna with no beam tilt or null fill in order to shoot over the water and not down onto it.)

Existing FM transmitter sites built on very tall towers in Riverview-Boyette showed good signals over an extremely wide area at night. This site is located 10 miles inland on the far side of the bay from our major cities. The received signals showed morning and afternoon fades, often 10 to 20 db, in the local metro areas beyond the bay, 20 to 40 miles away. These fades were most severe for stations operating with antennas over 1000 feet high. On the positive side, these signals showed very little short term signal fluctuations compared to signals radiated from the waters edge.

Existing FM stations operating at 500 to 600 feet and built several miles inland but near the center of population seemed to have the best all around signals, particularly during morning and afternoon drive times. They are subject, to a lesser degree, to the same kinds of over-water signal losses that the other stations experienced. These stations tended to have low gain antennas and high power transmitters.

CONCLUSION

This research enabled us to locate the optimum site for broadcast signal transmission to our four county Tampa Bay metro area.

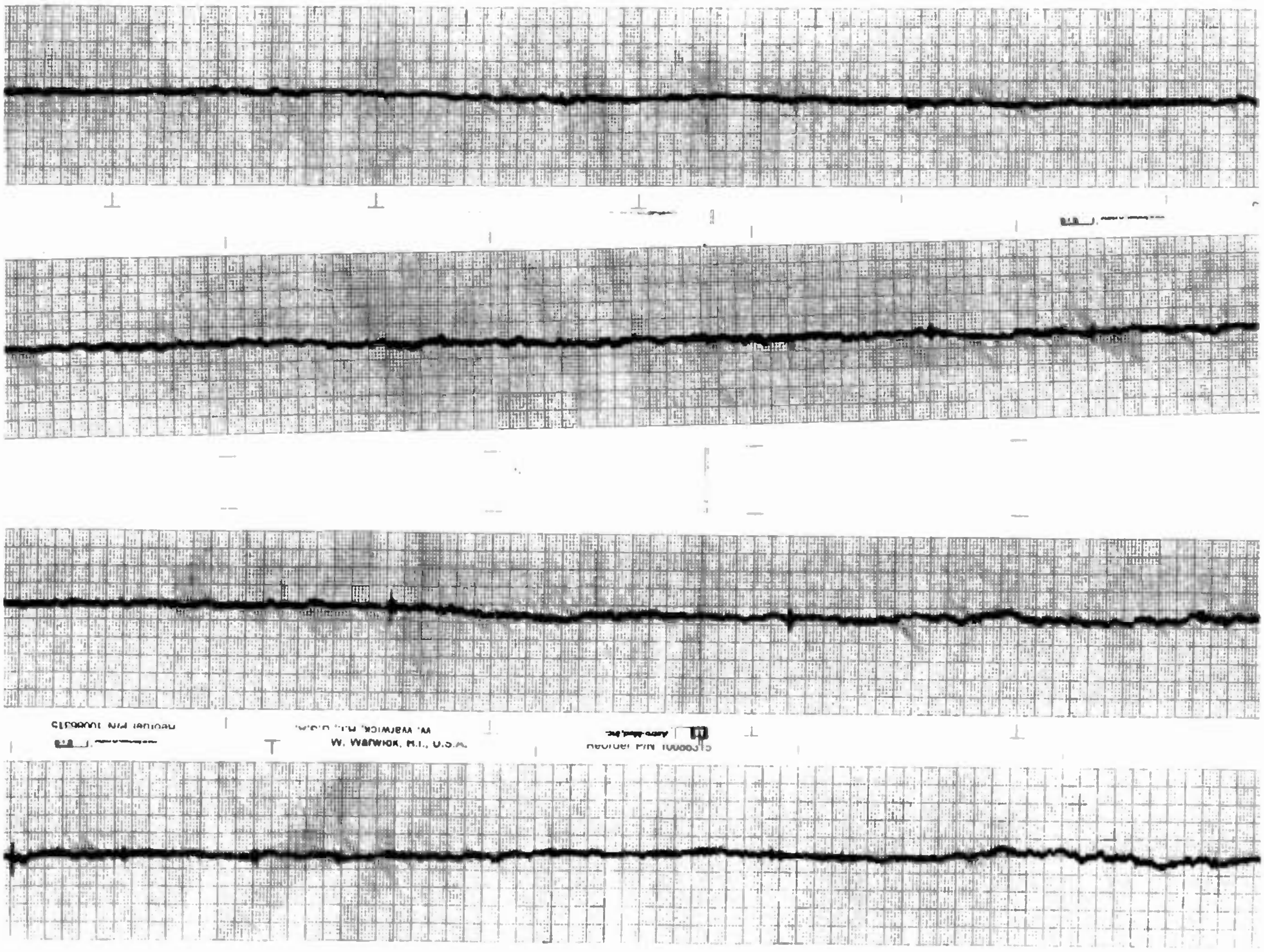
In our case, this site is located a couple of miles above the north central edge of the bay. The allowable tower height of 649 feet will keep us below the vast majority of temperature inversions. From this new tower we will have stabile land based propagation to the majority of the metro area population on the east, west and north sides of the bay. The distance to the remaining 25% of the metro population south of the bay is near enough that the signal includes more than 10db fade allowance before the 1mv signal level is reached.

This site also allows us to deliver "city grade" 3.16 mv signal to 90% of the four county metro population,(although no city grade signal is sent out into the ADI or TSA). This extra "in town" signal will help to hide the common effects of FM multipath. It will also help to guard against interference from semi-distant co-channel and first adjacent stations during coastal tropospheric ducting occurrences.

SUMMARY

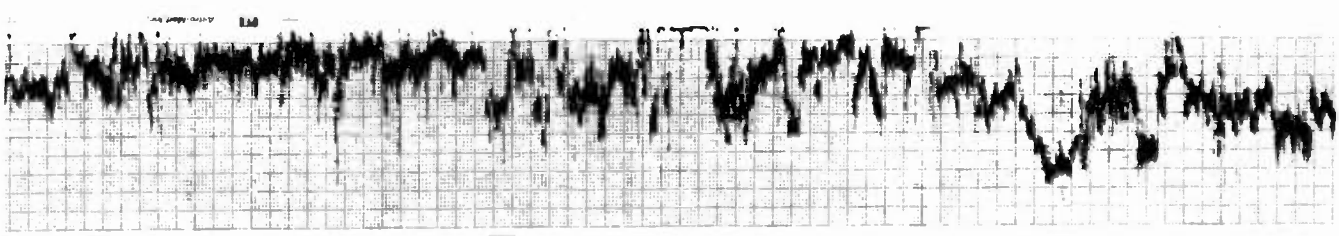
Our site selection choice was easy to make after carrying out careful field signal studies and correlating it to the geographical and meteorological features of our intended service area. In effect, we have learned how to control our signal after it leaves our antenna.

A broadcaster relying solely on the FCC 50/50 charts to guarantee a strong stable and competitive signal to his intended service area could be making a very serious and costly error if local conditions are not fully investigated prior to final tower site selection.

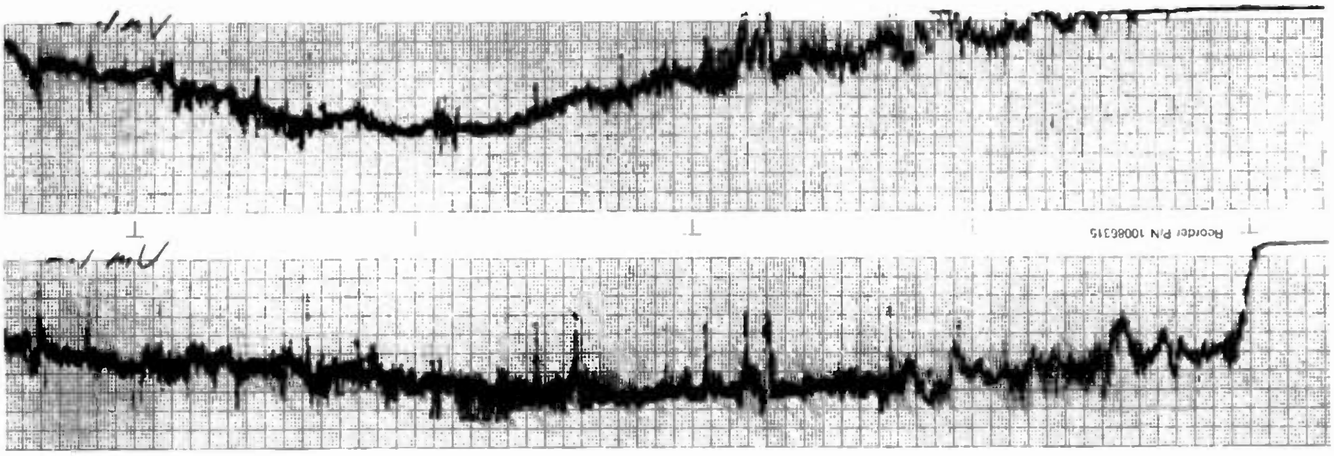


"NORMAL" SIGNAL PROPAGATION, SAMPLES ARE FROM THROUGHOUT THE DAY

MAXIMUM FLUCTUATION IS FROM 4.5 to 6 mv (2.5 db)



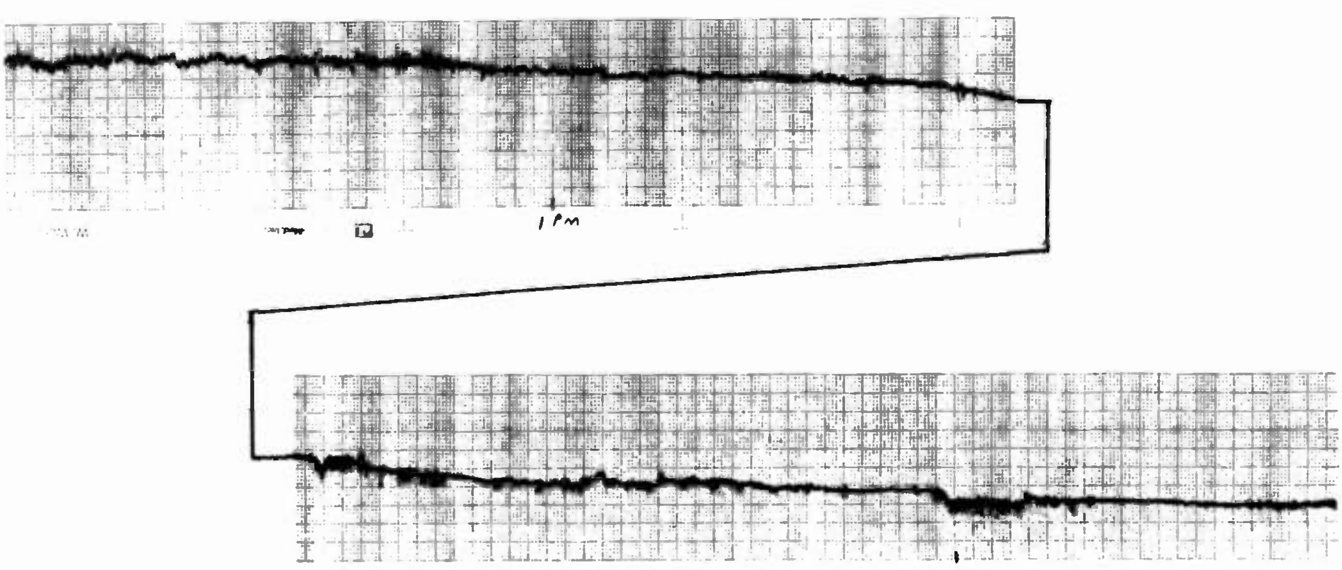
TRACE SHOWING THE EFFECTS OF BAD SIGNAL PROPAGATION



DUAL TRACE OF A BAD AFTERNOON FADE

IT AFFECTED THE SIGNAL ON THE UPPER TRACK BY 24 DB
THE LOWER TRACK SHOWED 16 DB OF LOSS.

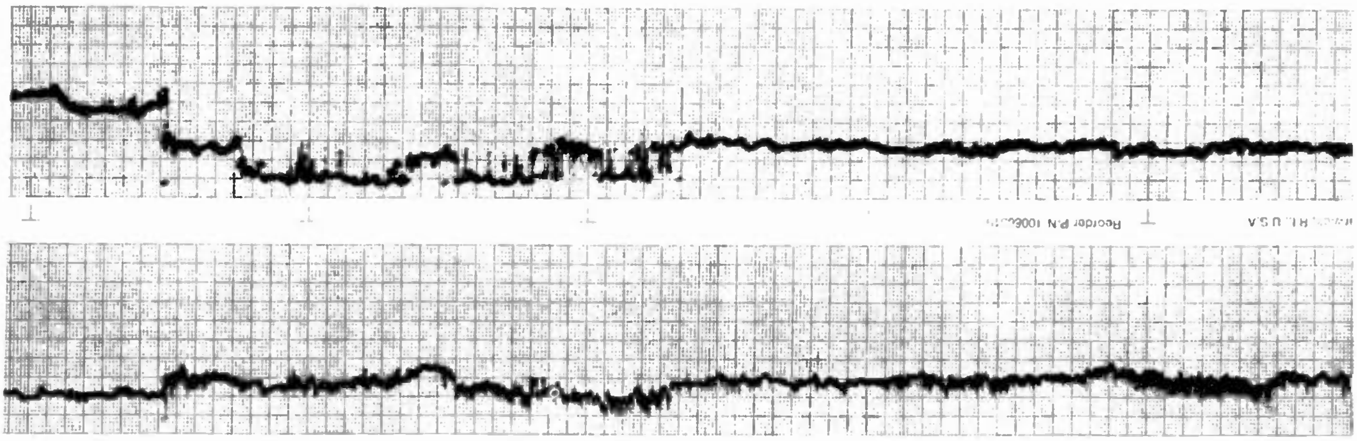
THIS FADE WAS SO DEEP THAT THE RANGE SWITCH ON BOTH FIM-71's
HAD TO BE INCREASED BY A FACTOR OF 10 (20 DB)



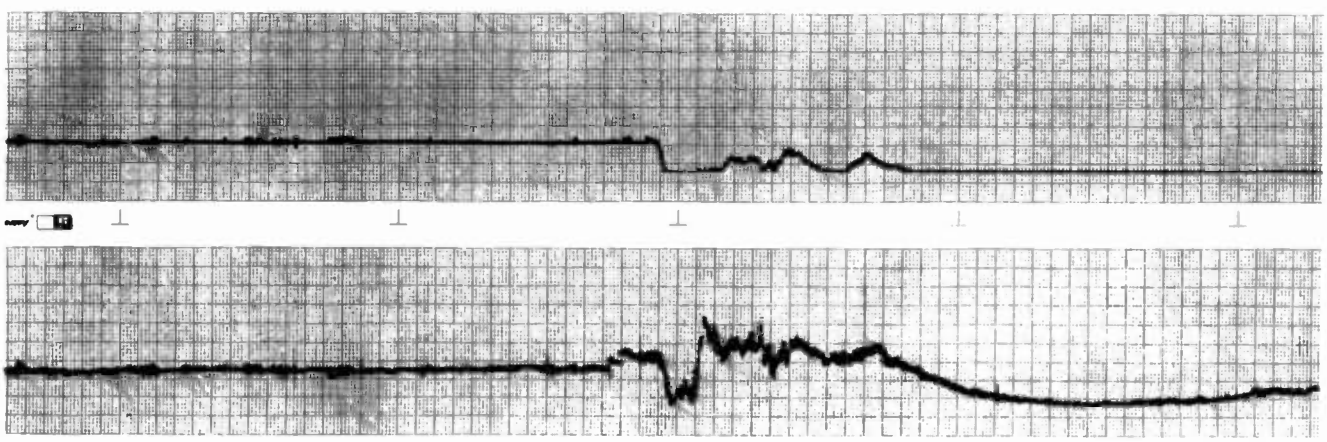
SINGLE TRACE SHOWING TYPICAL AFTERNOON FADE.

THE SIGNAL PEAKED AT 12:45 PM AT 8 mv,
AND DIPPED TO 2.2 mv AT 2:45 PM.

THE FADE IS CALCULATED AT 11.2 DB



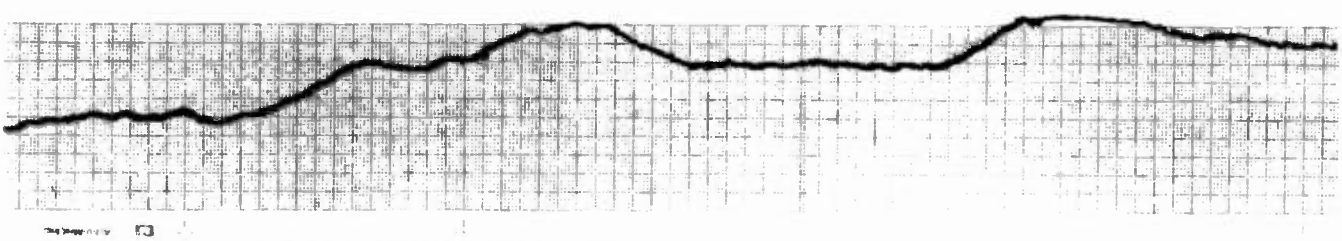
DUAL TRACE SHOWING THE EFFECTS OF MORNING FADES ON TWO SEPARATE STATIONS



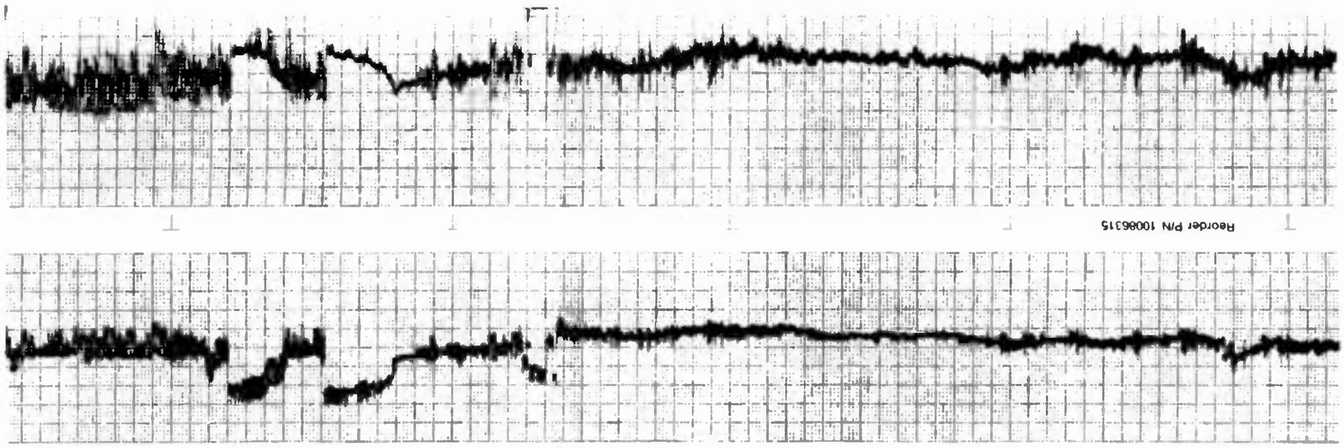
DUAL TRACE OF THE BEGINNING OF A 6 DB MORNING INVERSION LOSS.

UPPER TRACK IS THE CARRIER FROM A STATION TRANSMITTING FROM 1300 FEET HAAT.
LOWER TRACE IS FROM A STATION TRANSMITTING FROM 900 FEET HAAT.

NOTICE THAT THE LOWER STATION RECEIVED NEARLY 6 DB OF SIGNAL ENHANCEMENT
DURING PART OF THE INVERSION, AND STARTED TO RECOVERED QUICKER.

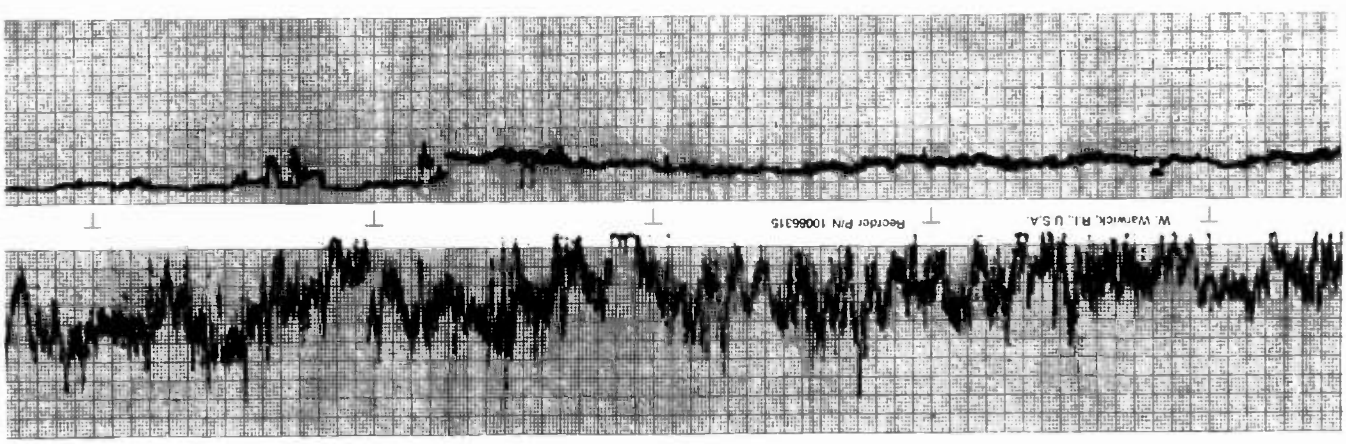


TRACE SHOWING THE END OF A 7.3 DB SIGNAL LOSS INVERSION



DUAL TRACE SHOWING THREE SHORT DURATION ANOMALIES, PROBABLY TEMPERATURE INVERSIONS NEAR THE 1000' LEVEL

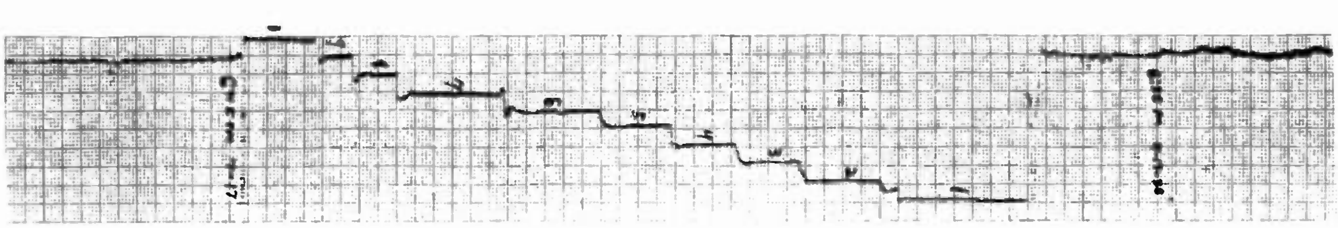
NOTICE HOW ONE STATION GAINS FROM THE INVERSION WHILE THE OTHER LOOSES



DUAL TRACE OF PROPAGATION ON A SUMMER AFTERNOON

UPPER TRACE IS AFTER 20 MILES OF OVER LAND PROPAGATION

LOWER TRACE IS AFTER 10 MILES OF WATER BASED PROPAGATION



SAMPLE OF CALIBRATION TRACE. NUMBERS WRITTEN ON THE CHART CORRESPOND TO THE METER READINGS ON THE FIM-71

OPTIMIZED IMPLEMENTATION OF SCA SUBCARRIERS FOR MINIMUM DEGRADATION OF FM STEREO RECEPTION

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ABSTRACT

Multiplex transmission systems have been used almost since the inception of FM broadcasting. Subcarriers can be added to the composite baseband modulating signal for the transmission of a variety of non-broadcast services and specialized applications. Transmission system performance requirements have been well established in order to provide minimal degradation of broadcast program service due to crosstalk from subcarriers. Unfortunately, additional factors beyond the control of the broadcaster can significantly influence the quality of reception. Most notable of these external contributors to impaired reception are consumer receiver characteristics (especially the stereo decoder circuitry) and multipath propagation. Care in the selection of subcarrier operating frequencies and methods of modulation can result in the optimization of both measured and perceived performance characteristics.

This paper reviews the history and current status of SCA technology and explores the general case of problem sources and their possible solutions. A specific case history is also described where unacceptable levels of performance were experienced when a conventional SCA service was attempted. An easy solution was discovered which provides an acceptable level of subjective and measured performance which was achieved by a careful analysis of the problem and utilization of a non-standard subcarrier frequency.

INTRODUCTION

FM Multiplexing: A Brief History

The first practical application of FM multiplex technology was achieved by Edwin H. Armstrong, the inventor of practical frequency modulation, when he demonstrated a four-channel transmission of high quality audio and facsimile via a single frequency-modulated transmitter from the Empire State Building over fifty years ago.[1,2] After World War II, when the FCC allocated VHF spectrum for television, the present FM band was established. The growth of FM was a very slow and painful process for the few pioneer broadcasters of the new medium, and the initial lack of receivers in the hands of the public hindered the economic viability of the new high fidelity service. A number of licensees turned to SCA revenues in order to remain solvent.

The introduction of stereophonic records to the listening public inspired the development of FM stereo, and a number of competing transmission systems were considered by the FCC. A "compatible" system developed by Zenith and General Electric was adopted in 1961. The FCC was aware of the reliance of many broadcasters on SCA revenues, and took this non-technical economic condition into consideration in making their selection of an FM stereo standard. They admitted that their field tests did not yield completely acceptable results with regard to crosstalk or interference between SCA subcarrier signals and stereo reception, and thus the "birdie" was born.[3] SCA operation with monophonic FM broadcasting had been virtually trouble-free, but the introduction of FM stereo (itself a form of multiplexing), which uses an amplitude-modulated subcarrier, created an entirely different animal.

SCA-To-Stereo Crosstalk

In order to protect the baseband spectrum occupied by the FM stereo signal, the FCC modified its rules regarding SCA multiplexing to specify that "the instantaneous frequency of SCA subcarriers shall at all times be within the range 53 to 75 kilocycles [per second]."[4] The industry adopted a de facto standard of 67 kHz, frequency-modulated, and the familiar 9 or 10 kHz birdie was created. Receiver manufacturers were confronted with this problem immediately and 67 kHz notch filters became standard equipment in virtually all FM stereo sets. A number of early studies of the problem correctly analyzed and defined the mechanisms involved and offered various suggested remedies.[5,6,7] The idea that the "high fidelity" stations should abandon dynamic range for compressed and maximized program modulation to mask the interference was offered at least as late as the 1970's.[8] Other factors not related to SCA interference contributed to the trend of "loud" FM stations that still prevails today.

With the improvement of receiver stereo decoder circuits in recent years, much of the problem with birdies has been greatly improved, but not eliminated. The wide variety of receiver designs and other factors such as the transmitter to receiver propagation path contribute to a wide variation in the severity of the problem.

Baseband and Modulation Limit Extensions

Recently, [in large part due to the work of John Kean while at National Public Radio[9]] the FCC modified the FM technical standards to allow a greater occupied bandwidth and higher total modulation level when SCA subcarriers are used. The baseband upper limit was extended to 99 kHz, allowing more than one subcarrier to be used simultaneously, and with a new total modulation limit of 110% (82.5 kHz peak deviation) when stereo and SCA subcarriers totalling 20% injection are employed.

The FCC rules have been relaxed regarding permissible modes of modulation which may be employed with subcarriers. Previously, FM was the only type of modulation allowed. The FCC now permits virtually any type or combination of modulation modes to be employed with subcarriers, so long as bandwidth limitations are not exceeded. Single subcarriers below 75 kHz may be used at up to 20% injection, or up to 10% injection level in the range of 75 to 99 kHz, although the FCC has granted authority for values above these limits upon a showing of compliance with the occupied bandwidth provisions of FCC §73.317. An example of such an exception is a high speed data subcarrier operating at 18% injection at a frequency of 79.5 kHz.[10]

The new freedom of choice for subcarrier frequency options and modulation methods has rendered the old 67 kHz receiver traps obsolete. Happily, many modern receivers perform well without the inclusion of such filters, and in such cases, the inclusion of a specific filter for subcarrier rejection will only impair other receiver performance parameters, such as stereo separation.

GENERAL SUBCARRIER CHARACTERISTICS AND PERFORMANCE REQUIREMENTS

SCA ("Subsidiary Communications Authorization") subcarriers for multiplex transmission have been used by FM broadcast stations for a wide variety of applications including background music, data distribution, remote control telemetry and paging. The application of this technology is only limited by the imagination of the user, the technical constraints of bandwidth and signal-to-noise ratio, and the level of impairment to and from the main program channel that is considered tolerable or acceptable. This last criterion is more subjective than measureable in engineering terms, because stations with agressive "loud" formats may be able to suffer considerable crosstalk in measurable terms without it becoming objectionably audible. Stations with classical music or other wide dynamic range formats may find any level of measureable or audible impairment of program quality to be unacceptable. Any "tune-out factors" or complaints from the listening audience are sure to weigh heavily against the potential benefits of subcarriers.

Since multiplexing by its definition is a "shared medium" technology, the degree of isolation or separation of the various simultaneous elements is never absolute, and the performance characteristics of each series element in the transmission path must be individually optimized for best overall crosstalk performance. Serious impairment can occur to the reception of the broadcast program channel or the subcarrier channel due to deficiencies at any point in the transmission path, including transmitting and receiving antennas, the propagation path in between, and the characteristics of the receiver employed, in addition to the integrity of the signal generated by the transmitter.

INTERMODULATION AND "BIRDIE" NOISE

The most common problem with regard to impairment of the FM stereo program channel is intermodulation distortion between the various components of the composite baseband signal that is present at the output of the FM detector circuit in the receiver. The result is a "birdie" noise, typically a high-pitched (in the vicinity of 9 or 10 kHz in the case of a 67 kHz subcarrier) that is modulated in pitch by the modulation that is present on the subcarrier (frequency modulation in the classical case).

Donald J. Popp studied this problem at Zenith Radio Corporation and presented a very thorough analysis at the 1963 AES convention.[11] His paper illustrates that the developers of FM stereo and receiver manufacturers had a very good understanding of the nature of SCA interference when FM stereo stations numbered only about 250, and more than a third of these stations were also using SCA subcarriers. Popp identified a number of receiver characteristics which contribute to birdie generation which will be discussed later in this paper.

The general findings of Popp's research are summarized here regarding the properties of SCA interference to stereo reception:

1. It is independent of stereo channel modulation levels.
2. It is the result of both second and third order intermodulation; therefore, two audio interference components are present simultaneously. The two prominent IM components formed can be identified as S-P and S-2P, where S is any instantaneous frequency of the SCA subcarrier and P is the frequency of the stereo pilot subcarrier [which is constant at 19 kHz].
3. It is dependent on the modulation of the SCA subcarrier for its character. [the distinctive "sound" of the interference]
4. It is dependent on the frequency deviation of the SCA subcarrier for its audio range.
5. The amplitude response follows the receiver de-emphasis curve.
6. The amplitude varies with the level of the SCA subcarrier [injection level].
7. When the IM products S-P and S-2P are presented to the 38 kHz [double sideband] stereo decoder circuit in a receiver, direct amplitude demodulation takes place, which produces the audible SCA interference.

Extending the analysis to the general case for all possible intermodulation products from all sources, the result is a series of frequencies defined by the expression:

$$n(P) \pm m(S)$$

where n and m are the harmonic orders of the pilot and SCA subcarriers. In most cases, the harmonics of the subcarrier are not significant and can be ignored. As a result, there exist exactly two significant audible interference products for every case where a single SCA subcarrier is employed. (For purposes of this analysis we will consider all frequencies less than or equal to 19 kHz to be audible.)

The expression: S ± nP produces the following series of significant IM products:

S+19, S-19, S+38, S-38, S+57, S-57, S+76, S-76,....etc.

For a 67 kHz subcarrier, the series becomes:

9, 10, 28, 29, 47, 48, 66, 67, 85, 86,....etc.

The stereo decoder direct demodulation of these terms (by subtracting 38 kHz from each) results in:

29, 28, 10, 9, 9, 10, 28, 29, 47, 48,....etc.

Of these products appearing at the left and right channel receiver outputs, only 9 kHz and 10 kHz are in the audible range, and are the resultant SCA "birdies".

For a 92 kHz subcarrier the S ± nP series is:

3, 16, 22, 35, 41, 54, 60, 73, 79, 92, 98, 111,....etc.

By subtracting 38 kHz in the stereo decoder we get:

35, 22, 16, 3, 3, 16, 22, 35, 41, 54, 60, 73,....etc.

Again, only two products are audible, 3 kHz and 16 kHz.

For the special cases where the subcarrier frequency is equal to a harmonic frequency of the pilot, and therefore S = nP where n is the harmonic order that is equal to the subcarrier frequency, the only IM products that result are a "zero beat" and the pilot harmonics. Therefore, the only "audible" terms become "zero Hz" and 19 kHz.

Because of the "zero beat" effect of SCA subcarriers on exactly 57, 76 or 95 kHz, this raises the possibility of an "interference-free" subcarrier, until modulation is applied.

NON-LINEAR DISTORTION SOURCES AND SOLUTIONS

In the preceding discussion of intermodulation and birdies, we analyzed the products of harmonic distortion from the "static" components of the composite baseband signal, since any non-linear distortion generally produces both types of products. Non-linear distortion is present in all FM transmissions to some degree, because the finite bandwidth limitations of the transmitter, antenna, receiver I-F stages, etc. remove "non-significant" higher order sidebands and alter the phase and amplitude of significant lower order sidebands. When a distorted frequency-modulated signal is demodulated, intermodulation distortion products are produced. As was previously described, crosstalk can occur when resulting IM products occupy frequencies within the baseband spectrum of the main channel, stereo subchannel or other subchannels.

There are three generalized areas where distortion can be introduced in the signal path:

1. The transmission system
2. The receiving system
3. The propagation path between 1. and 2.

Transmission System Performance Requirements

For a comprehensive analysis of the required level of performance of all the elements of the transmission system, see Mendenhall's paper from the 1986 NAB Engineering Conference.[12] The most important parameters involved are modulator (exciter) linearity, bandwidth limitations and asymmetry in all transmitter stages, incidental (synchronous) AM characteristics, and antenna performance (VSWR, asymmetry and bandwidth). Distortion products can be produced in the demodulated baseband signal by deficiencies in any of these areas.

Receiver Characteristics

RF, IF and Detector Stages

Distortion products are produced in the detected output of an FM receiver if any one of the following conditions exist:

1. Misoriented or inadequate receiving antenna contributing to substantial amounts of "multipath" reception where multiple signals of differing time delays, amplitude and phase are all present at the receiver input terminals simultaneously.
2. Mistuning of the receiver local oscillator such that the desired received signal is not centered in the passband.
3. Misalignment of IF or selectivity stages causing insufficient bandwidth or bandwidth asymmetry. This condition creates non-linear phase response and truncation of significant sidebands.
4. An FM detector stage with inadequate bandwidth or poor linearity.

In a properly designed receiver that is correctly tuned to a signal that is of sufficient amplitude and free of multipath echos and interference from adjacent or co-channel signals, none of the above factors is generally significant.

Stereo Decoder Circuitry

There are several methods of generating the stereo composite baseband signal and just as many ways of decoding the composite signal into the left and right channel components at the receiver. With the addition of SCA subcarriers to the composite signal, the method employed for stereo decoding becomes important, because the SCA signals are also "processed" by the stereo decoding circuit, resulting in significant products which may be in the audible range. This is a major area of concern and the most common reason for the wide variation in SCA crosstalk performance of different receiver designs.

Figure 1. is an early typical stereo decoder circuit of the diode bridge type.[13] The 38 kHz signal developed by doubling the pilot frequency is used to drive the diode matrix and the biased diodes are switched on and off on alternate half-cycles of the waveform. Because of the non-linear switching characteristics of the diodes and the distorted switching waveform, this circuit is very susceptible to interference from signals in the 57 to 95 kHz region. The presence of a significant 76 kHz component in the switching waveform results

TYPICAL EARLY STEREO DECODER CIRCUIT
WITH 67 KHZ INPUT FILTER

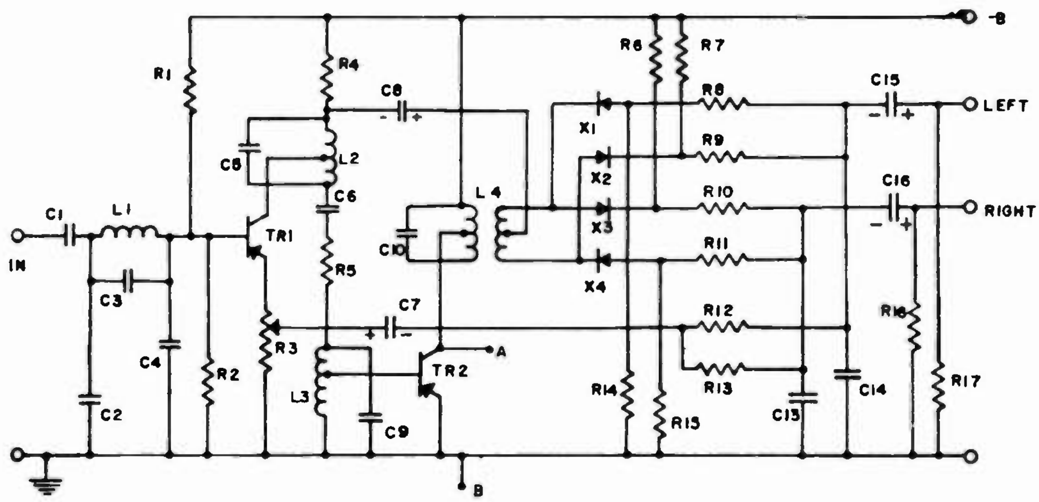


FIGURE 1

in the direct demodulation at this frequency. A 67 kHz subcarrier produces a large 9 kHz component at the outputs (76 - 67 = 9 kHz). An input notch filter attempts to suppress the 67 kHz signal, and the de-emphasis curve provides some additional attenuation, but the resultant birdie is usually very audible. The 67 kHz input filter also degrades the stereo separation due to the phase distortion that it introduces in the region of the L-R upper sideband. Abandonment of this type of circuit in recent years has been the single most significant improvement in SCA to stereo crosstalk performance in modern receivers.

Modern receivers almost universally utilize an integrated circuit switching detector that uses a square wave switching waveform. The 38 kHz square wave switching waveform is usually derived from a 76 kHz voltage-controlled oscillator that is phase referenced to the 19 kHz pilot by a phase-locked loop circuit. A well-executed design will have a square wave with nearly perfect symmetry (50% duty cycle) and no even-order harmonics. Such decoders show good performance in the 57-95 kHz region since there is no 76 kHz switching component to cause demodulation in the audible range from subcarrier signals.

Unfortunately, a true square wave has a significant third harmonic component (and less significant higher odd-order harmonics), and therefore is not an "ideal" switching waveform. The result is direct demodulation at 114 kHz which can produce audible components if signals are present in the 95 to 133 kHz region. This is the so-called 114 kHz "taboo" that is the main reason behind the FCC baseband limit of 99 kHz, since signals in the 99 to 114 kHz region produce birdies at 15 kHz and below. One solution to this problem that has been successfully implemented is a cancellation circuit that uses an additional switching detector driven by a 114 kHz square wave such that the outputs of the two detectors are combined out-of-phase, thus cancelling the birdie noise.[14]

Other switching waveforms can be used with excellent results, so long as the characteristics of the waveform are known and precise. For example, a triangle wave has less harmonic content than a square wave and therefore might be a good choice. A pure sine / cosine wave would most likely be an ideal waveform to drive a precision voltage-controlled attenuator circuit. One developmental proposal is to use a step-synthesized sinusoidal switch using switched-capacitors.[15] An even more exotic approach is to use digital processing techniques in the form of a "digital tanlock loop".[16] These designs confirm that there is considerable room for improvement in receiver "state-of-the-art".

Unfortunately, some receiver manufacturers are using circuits that have serious switching waveform distortion or asymmetry.[17] Sometimes this is a result of poor power supply decoupling of the 76 kHz PLL oscillator and the 76 kHz signal gets into "places where it doesn't belong". Direct switching demodulation at 76 kHz will always produce birdies as discussed previously. The recent manufacturing trend of putting most of the receiver circuitry on a single chip and the use of frequency synthesizers for discrete digital tuning and display, has in some cases made the problem worse, by introducing spurious signals into detector and stereo decoder circuits. Many digital displays use an oscillator-driven scanning circuit which can also crosstalk into RF, IF, and demodulation stages, compounding birdie problems if audible beat components are produced.

Actual measured receiver crosstalk performance data for several early receivers are tabulated in Popp's AES paper.[18] A similar study of receivers available to this author was undertaken and some of the results are presented in a later section of this paper.

Propagation Path Distortion

Distortion of frequency modulated signals due to propagation conditions has been studied at least as early as 1936 by Crosby[19] and others.[20] A rigorous analysis of the nature of harmonic distortion as related to sideband phase and cancellation due to multipath echoes was undertaken by these FM pioneers, and attempts at solutions to this problem through the use of diversity reception were also engaged. These early experiments were conducted at low VHF and short-wave frequencies where propagation conditions are somewhat different than at 100 MHz, however their findings are relevant to the general case.

Multipath distortion is the most prevalent and perplexing problem that contributes to SCA crosstalk. When multiple transmission paths are present with significant time delay and phase difference between the received signals, the detected FM signal can contain serious harmonic and IM distortion. When components from subcarrier presence fall in the passband of the stereo subchannel, birdies result. This effect is particularly noticeable in automobile receivers where the terrain effects and the moving receiver produce "picket fencing" noise, especially in an urban environment where large man-made structures with flat surfaces (notably office buildings) produce significant echoes of the direct RF signal.

A number of developmental solutions of the multipath problem have been proposed, including manual and automatic echo cancelling circuits.[21,22] In one design, a computer calculates the relative echo amplitudes, delay times and phase differences from outputs of an IF signal envelope detector and FM demodulator. Controlled delays in the form of a "programmable transversal filter" operating at the linear IF stage are adjusted to produce signals which are combined with the direct IF path to cancel the echoes. The major disadvantages of this system are the complexity and the slow rate at which it operates, thus it is not practical for a mobile receiver where the echo characteristics are continuously changing. Another approach that has been more successfully employed is a system of antenna/receiver diversity. In one case a special antenna consisting of three independent elements, two small crossed loops and a short monopole are employed to discriminate between direct and indirect signal paths.[23] This system is most effective when the direct and echo paths are coming from different directions. Unfortunately there are many cases where multiple reflections can arrive from the same direction as the direct signal.

Schober has shown that some multipath "noise" is due to incidental or synchronous amplitude modulation of the transmitted signal.[24] Optimizing the transmitter performance by careful monitoring and tuning of all transmitter stages for minimum incidental AM can make a noticeable improvement in the severity of the distortion produced under severe multipath conditions, where signal cancellation produces an extremely weak signal for the receiver to work with. Under such conditions, the limiting action is not complete and AM components in the signal become significant. The locations where deep nulls in signal strength occur produce areas where IM distortion is severe, and the addition of significant AM components worsens the problem. Figure 2 and Figure 3 taken from Schober's article illustrate a hypothetical case where significant improvement in the absence of incidental AM can be demonstrated.

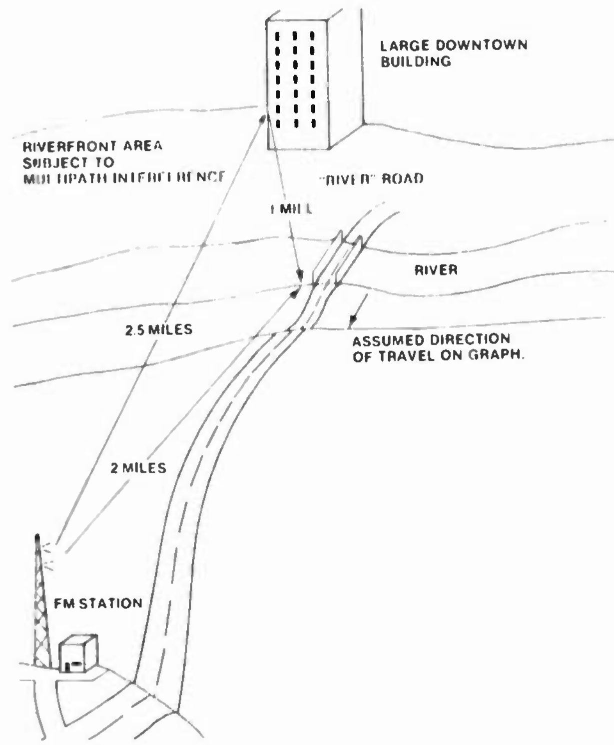


Figure 2. Hypothetical case of a situation generating field strength variations.

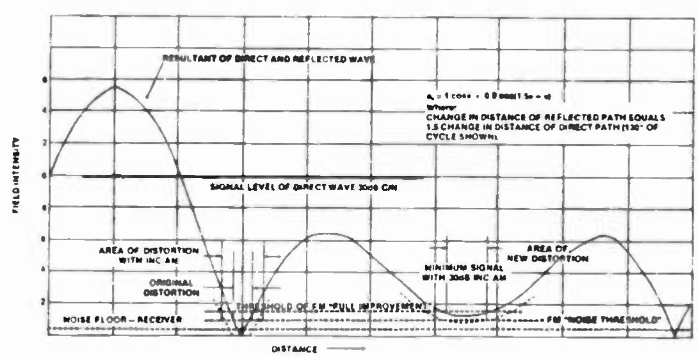


Figure 3. Field intensity vs. distance for two interfering signals, assumed in the situation shown in Figure 2.

An extensive computer-aided analysis of multipath distortion where particular attention was paid to SCA crosstalk was undertaken at NHK, Japan.[25] In this study they identified the second order IM component of S-P to be the predominant product in most cases, and the demodulated birdie product is:

S - P - 38, or S - 57. (67 kHz becomes 10 kHz)

Third order components are also present but at a much lower amplitude. In general, crosstalk and distortion increase with echo delay time, and the desired to undesired ratio necessary for acceptable levels of crosstalk also increases with delay time, and is on the order of 20-30 dB for delays of 5-20 microseconds. Their criterion for acceptable crosstalk was -55 dB, which can still be quite audible, and would not be considered acceptable to many broadcasters. Other studies have shown that the perceptible level of crosstalk can be lower than -65 dB[26] and under certain conditions as low as -74 dB.[27]

It is obvious that multipath must be minimized for acceptable levels of SCA crosstalk, which can best be achieved by optimizing the receiving antenna for the existing RF environment, a situation that is not always practical. The author has observed a seasonal variation in multipath conditions that can be attributed to a change in the "reverberant" characteristics of the terrain, due to such factors as deciduous forestation and snow cover. Line cord antennas and "rabbit ears" account for the familiar phenomena of distorted audio and "ghosty" TV pictures when you walk around the room, however, tidal forces and lunar phases are not expected to induce noticeable effects.

SCA IMPLEMENTATION AT A CLASSICAL MUSIC STATION: A CASE HISTORY

Background

The University of Michigan public radio stations were approached by a client who wished to lease a subcarrier channel. They would be responsible for the aquisition of suitable receivers which they would then lease to their clients. They also agreed to delivery of their program audio to the SCA generator. WUOM would provide the SCA generator and essentially lease the use of the channel it provided. A contract with a test period clause was drawn up and the expectation was substantial revenue at a minimum of investment, labor or distractions on our part. The initial attempt was not successful.

Conditional Test Period

After consulting with a number of industry people familiar with SCA utilization, we agreed to a test period with a conventional frequency-modulated SCA subcarrier at 92 kHz. WUOM had been using 67 kHz at a level of 3% injection for subaudible analog remote control telemetry for several years, and the characteristics of that mode of operation had become very familiar to us (including a birdie in the 10 kHz region that was most noticeable with a stereo automobile receiver under multipath or slightly "detuned" conditions). We kept the SCA turned off when we were not reading meters. Since WUOM is a fine arts classical music station that uses minimal audio processing and carries wide dynamic range program material including live concerts, we were cautious as to committing ourselves to any agreement which would involve any noticeable impairment of our stereo program signal.

As soon as the test period commenced, I immediately got complaints from staff announcers that something was wrong with our air monitor (a crystal-controlled single channel monitor receiver). They complained that they were hearing "funny noises" in their headphones during quiet musical passages, and when they were announcing. They admitted that the noise (which consisted of a modulated 3 kHz birdie) was "barely perceptible," but extremely annoying. I auditioned the off-air signal with several different receivers and found the same birdie, at various levels of annoyance. The measured level in the monitor receiver was better than 68 dB below 100% modulation, and actually below the noise floor of the composite STL, but it was loud enough to be noticed. The consensus of the staff was that it was a SERIOUS PROBLEM. Because of the variation of birdie levels in different receivers, I began to suspect that the PROBLEM was primarily related to the receiver designs, and noted that one particular receiver seemed to be virtually immune to this phenomenon.

Finding "The Problem"

When the SCA generator was installed, care was taken to insure proper operation, and the transmitter tuning was optimized for minimum incidental (or synchronous) AM, an important parameter recognized by several sources.[28,29,30,31,32] The transmitter plant had been completely replaced in 1980, including the tower and antenna, and a review of maintenance records and distortion measurements did not reveal even a hint of a "problem". The transmitter is a single-tube 25 kilowatt tetrode amplifier with broadband solid-state driver stages. The antenna is also very broadband and was field-tuned by the manufacturer with an admittance bridge. There was no mismatch between the exciter and the transmitter stages, and various lengths of interconnecting cable were tried with no effect. The 700 feet of 3-1/8" transmission line was not suspect, and this length is too short to have resonant modes at any frequency related to 92 kHz.

Spectral analysis of the composite baseband signal at the input of the exciter showed no spurious components and stereo generator pilot harmonics were well-suppressed. The composite output of the modulation monitor was not significantly different from the spectrum observed at the exciter input, so I ruled out exciter or transmitter deficiencies as the source of "The Problem".

Listener Complaints

By now, we had started to receive a few complaints from listeners, especially listeners to one of our "satellite" stations, WFUM-FM in Flint, Michigan.

WFUM-FM re-broadcasts the programming of the "mother station", WUOM (no pun intended), via an off-air monitor receiver from a distance of more than 40 miles. The received stereo signal is demodulated to left and right audio channels at the studio site of WFUM(TV). These stereo audic signals modulate FM subcarriers on the WFUM(TV) microwave STL. Audio noise reduction (compander type) is used over the STL circuit to improve the noise performance, and a peak limiter is used before the stereo generator located at the WFUM-FM (and TV) transmitter site.

The off-air relay receiver used by WFUM-FM is identical to the one used as an air monitor by our announcer staff at the WUOM studios. Since the audible birdie from the output of this receiver was being re-broadcast by the WFUM-FM transmission chain, it was audible in all stereo receivers, despite the fact that the SCA itself was not present on the WFUM-FM transmitted signal. It is notable that the birdie was completely inaudible on monaural receivers, despite the fact that the signal was definitely present in both left and right audio channels. This was due to the fact that the birdies were out-of-phase with each other (180°) in the left and right channels.

The Receiver Cure

When the WFUM-FM engineers discovered that with the replacement of their off-air receiver with a newer consumer-type FM tuner, their "problem" was "cured". This receiver "cure" was encouraging in that it meant that the WUOM signal was "clean", but very discouraging in that the performance of receivers in the hands of the listening public was unknown, and many of the receivers to which I had access exhibited unacceptable levels of birdie output. Automobile receivers of both older and modern design all seemed to exhibit the problem, especially in the presence of multipath.

The Birdie

The birdie which we experienced with the 92 kHz SCA was described to our general manager by a listener in this way: "It sounds like someone is sending Morse code." This is a very good description of the phenomenon, due to the nature of the modulation that was present on the subcarrier.

The SCA program consisted typically of an announcer reading text. During the pauses in his speech (between words and even between some syllables), the subcarrier was essentially unmodulated. The unmodulated subcarrier was beating with another signal that resulted in a 3 kHz tone that was interrupted when the subcarrier energy was dispersed into sidebands with modulation. Thus, the "holes" between speech sounds became "dits and dahs" that sounded to the untrained ears of a listener like Morse code.

By beating an external oscillator to the pitch of the birdie (like a piano tuner adjusts the pitch of a piano note) it was determined that the resting frequency of the unmodulated birdie was exactly 3 kHz, as measured on a frequency counter.

After doing some simple arithmetic, it was suspected that the 3 kHz was a product of 92 kHz and the fifth harmonic of the 19 kHz stereo pilot at 95 kHz. However, after spectral analysis of the outputs of a few receivers and a stereo modulation monitor, it was discovered that a significant amount of 16 kHz was also present. In the case of the modulation monitor without de-emphasis, there was more 16 kHz birdie than 3 kHz!

Adult human hearing does not extend to 16 kHz in most cases, as some high frequency impairment tends to progress with age. Young ears can detect that high a pitch (and higher) but with much less sensitivity than a mid-range pitch (such as 3 kHz) where the ear is most sensitive. Because of de-emphasis in FM receivers, a signal of equal modulation level at 16 kHz would be about 14 db down from a 3 kHz signal of equal amplitude (75 microsecond de-emphasis).

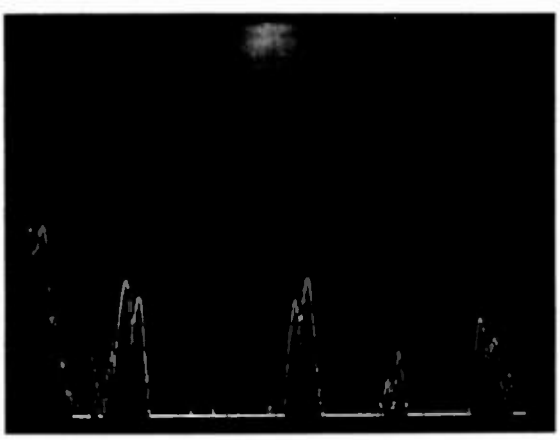
As illustrated in the section on IM and "birdie" noise, it can be shown that there are several possible mechanisms that can produce 3 kHz and 16 kHz products from a 92 kHz SCA. The most likely one in this case (in the absence of multipath) is caused by the 114 kHz demodulation action in most stereo decoders. If we subtract 92 from 114 we get a second order IM or demodulation product at 22 kHz. Subtracting 22 from 38 we get a third order product at 16 kHz. It is also possible to get a second order 16 kHz product if a 76 kHz switching component is present (92-76=16). This appeared to be a factor in the modulation monitor because it was noted that when the symmetry of the 38 kHz waveform is adjusted (there is a control provided inside the monitor for this alignment) the 16 kHz product can be nulled out. This null appeared to occur close to the adjustment that gave the most symmetrical looking waveform.

The mechanism for the 3 kHz component of the birdie was also most probably due to 114 kHz demodulation. If the IM product of the sum of the pilot and the subcarrier is significant (19 + 92 = 111 kHz), this will directly demodulate to 3 kHz when subtracted from 114 kHz. Other mechanisms are certainly possible.

Figure 4 is a spectrum analyzer photograph of the left channel output of a highly regarded FM tuner from a distance of about 12 miles from the WUOM transmitter. The receiving antenna was adjusted for minimum multipath (minimum IM distortion products) and the transmitter was modulated only by a 19 kHz simulated pilot at 10% and a 92 kHz simulated subcarrier at 10%. Two low-distortion oscillators were used instead of stereo and SCA generators in order to insure that the harmonic content of the modulating signals was not significant. The baseband spectrum from zero to 100 kHz is displayed. The residual pilot and its 57 kHz image due to 38 kHz demodulation are about 70 dB below the level of a 100% modulated low frequency signal. A low frequency reference was used because of the 75 microsecond de-emphasis in the receiver. Products are visible at 22 kHz and 54 kHz. Both are at a level of about -82 dB. The 54 kHz signal is the result of direct demodulation of the 92 kHz subcarrier at 38 kHz (92-38=54) and the 22 kHz signal is suspected to be due to direct demodulation at 114 kHz (114-92=22). All audible products are unmeasureable (better than 90 dB down).

0 Hz Figure 4 100 kHz
92 KHZ SCA LEFT CHANNEL RECEIVER OUT NO MULTIPATH

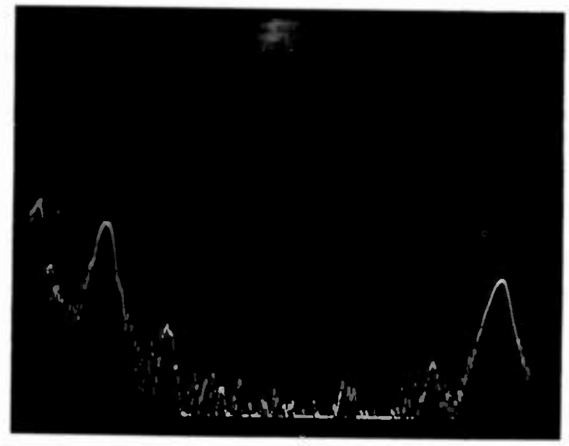
Figure 5 shows the output of the same receiver as Figure 4 (92 kHz SCA) but with multipath introduced by re-orienting the receiving antenna for maximum distortion. The 22 kHz and 54 kHz signals are now at about -74 dB (6 dB worse) but a 3 kHz birdie product has appeared that is at about -60 dB.



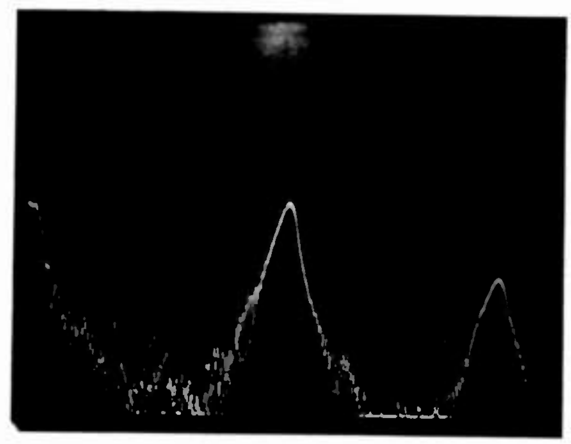
0 Hz Figure 5 100 kHz
92 KHZ SCA LEFT CHANNEL RECEIVER OUT WITH MULTIPATH

Figure 6 is the same output as Figure 5 except only the audible range is displayed (0 to 20 kHz). A 16 kHz product can be seen at about -86 dB. The spur at about 5.5 kHz down 80 dB is of unknown origin.

Figure 7 is the same receiver output and the same reception conditions as Figure 5 and Figure 6 but with a 67 kHz SCA. The 10 kHz birdie is at -55 dB.



0 Hz Figure 6 20 kHz
92 KHZ SCA LEFT CHANNEL RECEIVER OUT WITH MULTIPATH



0 Hz Figure 7 20 kHz
67 KHZ SCA LEFT CHANNEL RECEIVER OUT WITH MULTIPATH

92 KHz Performance Is Unacceptable

The previous spectral analysis of a high quality receiver gave a worst case birdie measurement of -60 dB under severe multipath conditions. To most stations operating with agressive formats, this would probably be tolerable. However, the consensus of the staff at WUOM expressed dismay at the obvious impairment to the signal, and management concurred with this sentiment (this WAS a PROBLEM).

Alternatives to 92 KHz

Not yet ready to give up, I hauled my trusty sine-wave sources out to the transmitter site, determined to find an alternative to 92 kHz. I decided to experiment using the real stereo generator this time to simplify the procedure, since I was confident now that the transmission system itself was in reasonably good shape (distortion and birdie products were being generated externally). I adjusted my oscillator to modulate the transmitter at about 20% this time (twice the normal injection level) to exaggerate whatever problems I would find. I had an assistant at the studio attempt to document our findings and he monitored our signal using two different receivers. I had my Walkman at the transmitter site. As I expected, as I swept my synthetic subcarrier up from the audible range, a zero beat was produced every 19 kHz (at each harmonic of the pilot). A subcarrier frequency that was 1 kHz either side of a pilot harmonic frequency produced a 1 kHz tone in all three reference receivers.

The subjective results were similar for all three receivers. The S - nP beat note that was produced as the S signal was swept through each pilot harmonic frequency differed substantially in amplitude depending upon which harmonic the S signal was beating with. Here are the results in order from loudest (worst) to quietest (best):

- 38 kHz (direct stereo demodulation)
- 114 kHz (illegal, exceeds permitted bandwidth)
- 57 kHz (legal but loud)
- 95 kHz (only about 2 dB better than 57)
- 19 kHz (illegal, in stereo baseband)
- 76 kHz (quietest)

The Discovery

This initial experiment produced results that looked promising. Not only had I discovered that subcarriers in the vicinity of 76 kHz caused the "quietest" birdies (in the three receivers that I initially tested), but the thought occurred to me that if it were possible to use a frequency of exactly 76 kHz, a true zero beat would be present when the subcarrier was unmodulated. It also became obvious that the only possible IM distortion products were pilot harmonics and zero beats, none of which seemed likely to cause reception problems.

Looking back at our experience with 92 kHz, I pondered the data obtained and considered why the "problem" had been insurmountable. Although the numbers looked pretty good, the bottom line that sealed the fate of 92 kHz was the perceived or subjective performance, not the measured performance. If a data subcarrier had been implemented, perhaps the results would have been successful, since the continuously dispersed modulation of the subcarrier would not have produced a discrete tone. This "tonal" quality of the interference and the fact that the resultant audible pitch was in the most sensitive spectrum of human hearing certainly weighed the evidence against using 92 khz at WUOM.

The Second Experiment

Plans were set in motion to modify the SCA generator for 76 kHz, and the manufacturer was contacted to attempt to use this frequency with the modulation conditions that the SCA client was providing. The manufacturer had not tried 76 kHz before, and it took them a few days to come up with the modifications required, which were quite simple and took about an hour to accomplish. The circuit change required the addition of a single capacitor and the changing of a few wirewrap connections. The programmable subcarrier frequency determining circuit in the generator allowed discrete steps of adjustment, and the closest frequency they could come up with was a nominal 75.992 kHz. The preferred technique would be to phase-lock the rest frequency of the SCA modulated oscillator to the 19 kHz pilot, but that would have required extensive modification of the SCA generator. Since the FCC pilot frequency specification is 19 kHz ± 2 Hz, the calculated beat frequency would be less than 16 Hz. If necessary the frequency of either the stereo generator or SCA generator could be trimmed slightly to bring the beat frequency closer to zero.

Since the client already had 92 kHz receivers in the field and was busy trying to lease them to customers, it was necessary to test the 76 kHz SCA on a weekend, and then put things back on 92 kHz until the client could obtain or modify his receivers for 76 kHz. This didn't give us much time to experiment.

The initial results were very encouraging. With or without modulation, our announcers could not hear any birdies with an injection level of 10%. Automobile receivers didn't seem to care about the SCA at 76 kHz either, although the subjective effect was a more pronounced picket-fencing noise. However, the additional noise was completely atonal in nature (dispersed "white" type noise) and thus no new harmonic dissonance was introduced into the musical main channel program. The 3 kHz birdie from the 92 kHz subcarrier was most offensive when quiet musical passages were broadcast, since the interference was always "out of tune" with the program.

A second weekend of testing at 76 kHz revealed no new flaws. This test was necessary to see if the client could retune his existing receivers for his customers. This procedure was readily accomplished by tuning the receivers to the WUOM broadcast signal (SCA phase-locked loop adjustment). A receiver "swap" was arranged and WUOM commenced 76 kHz SCA service on a conditional basis (contractual test period).

New Listener Complaints

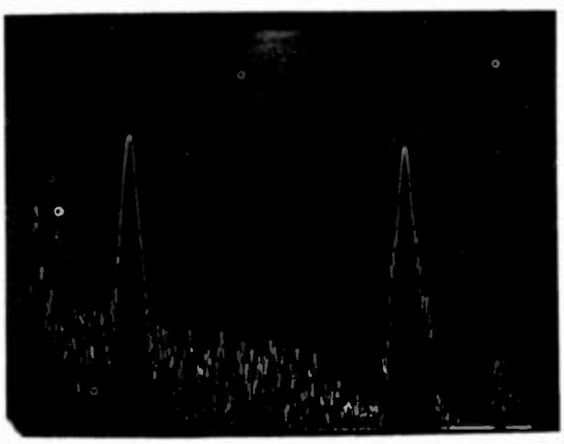
Just when we thought we could have our cake and.... (I'll try to spare the cliché's) I received two complaints from listeners about a low-frequency growling noise on our signal. They also described a very distorted "monkey chatter" sort of interference, that could be identified as an unintelligible human voice. None of the receivers I had tried at this point had a significant level of crosstalk. Both receivers in question turned out to be more than ten years old, and each had a stereo decoder circuit very similar to Figure 1.

Due to the nature of the prevailing stereo receiver circuits (and the old FCC baseband limitation of 75 kHz) it became obvious why 76 kHz had not become a viable SCA frequency in the early days of stereo FM broadcasting. Fortunately receivers have been improved drastically since the introduction of an economical square-wave switching decoder (in the form of a wide variety of PLL integrated circuits).

WUOM has received a total of three (3) complaints since 76 kHz SCA operation began in mid-1986. The population of diode-matrix stereo receivers has diminished substantially, and most current model receivers exhibit good crosstalk performance with 76 kHz, including several very inexpensive radios tested. The few exceptions (mentioned in the receiver section of this paper) that have come to my attention exhibit crosstalk performance that is not ideal for critical applications, but would be more than adequate for stations with agressive modulation.

76 KHz SCA Performance

Figure 8 is a spectrum analyzer display of the transmitted WUOM baseband signal sampled from a receiver with a composite output (thanks to the quadriphonic debacle of the 1970's). Note the S + P intermodulation product at 95 kHz that is about 66 dB below 100% modulation.



0 Hz Figure 8 100 kHz

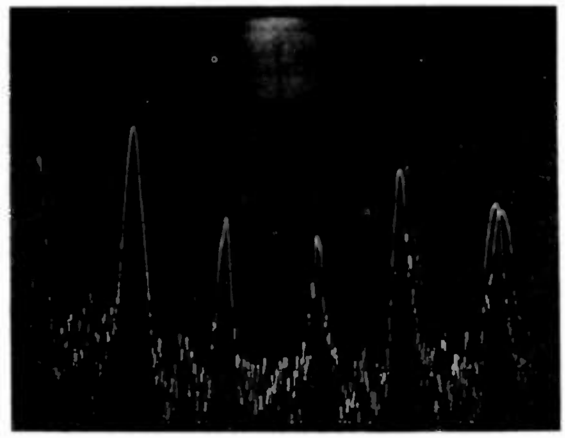
76 KHZ SCA WUOM ANN ARBOR, MICHIGAN NO MULTIPATH

Figure 9 is the same received signal with the antenna reoriented for maximum multipath distortion. IM products are S + P, S - P, and S - 2P.

CONCLUSIONS

The FM composite baseband spectrum from 53 to 99 kHz is available for SCA subcarrier use. The distinct advantages of 76 kHz when a single frequency-modulated subcarrier is contemplated have been demonstrated in theory and practice. Wideband or even "double multiplex" applications may prove practical, especially as receiver designs improve, as 76 kHz is exactly centered in the designated SCA spectrum.

Additional subcarriers can be added at 57 and 95



0 Hz Figure 9 100 kHz

76 KHZ SCA WUOM ANN ARBOR, MICHIGAN WITH MULTIPATH

kHz without interference to 76 kHz operation if bandwidth is reduced. In particular, RDS or ARI or similar synchronous 57 kHz subcarrier systems can be accomodated without interference to or from a 76 kHz SCA. Even with subcarriers at both 57 and 95 kHz, the available bandwidth at 76 kHz is greater than that normally utilized at 67 kHz. A full 38 kHz bandwidth is avilable at 76 kHz if single-sideband subcarriers are used at 57 and 95 kHz. The zero-beat properties of 76 kHz greatly improve the subjective crosstalk performance with stereo program service, especially in the presence of multipath-induced intermodulation.

Receiver manufacturers should incorporate designs that optimize crosstalk performance throughout the 53-99 kHz SCA baseband spectrum, as this entire spectrum will be utilized in an increasingly multifarious way, as new applications and techniques for SCA technology are envisioned and developed.

ACKNOWLEDGMENTS

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National Association of

NAB

BROADCASTERS

A PROPOSAL FOR INCREASING POWER OF CLASS A FM STATIONS

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ABSTRACT

Research work from 1983 to present shows that Class A FM Broadcast stations may increase the ERP from 3 kW to 6 kW with no adverse effects on the allocation spectrum and provide meaningful improvement to the service area:

1. Minimize the disparity between competing facilities in markets with mixed classes of station (A and C or A and B).

2. Increase penetration of buildings in urbanized areas while overcoming terrain shadowing with higher antennas.

3. Minimize the increased adjacent channel interference during periods of atmospheric inversions.

4. Increase coverage within rural areas without requiring prohibitive capital investment and lengthy legal proceedings.

HISTORY

The research work on improving Class A stations began in the summer of 1983 when Clear Channel Communications Inc. purchased KPEZ, a Class A FM station in Austin, Texas. This was a unique opportunity as KPEZ was then located on the same tower with KHFI, another Class A station in Austin. When the station was purchased, it was operating with 152 meters (500 feet) HAAT and 700 watts ERP. KHFI was operating 125 meters (410 feet) HAAT and ERP of 1,500 watts.

There will always be listener complaints given to a Class A station operating in a metropolitan area competing with Class B or Class C stations. We found, however, that there were numerous complaints from listeners in commercial buildings in the business district. Tests with a Potomac FIM-71 showed that KHFI fared better in the buildings than KPEZ. The easiest answer was to reduce height and increase ERP to penetrate the buildings. Because the antenna space at 125 meters was occupied by KHFI, we relocated our antenna at 91 meters (300 feet) to achieve maximum ERP (the transmitter site is within 199 miles of Mexican border). This effort did work, both from measurements and from listener response.

The Austin, Texas, area is mixed flat land and hill country. The reduction of height to 91 meters caused complications in the areas of irregular terrain, mainly dropout and multipath. KHFI fared better in these areas and also in fringe areas because of increased height. Since the move to 91 meters, KHFI vacated the tower and KPEZ has a C.P. to move to 125 meters HAAT.

Another complaint received occasionally was 1st adjacent channel interference with temperature inversions during season changes. The interfering station is operating with 524 meters (1720 ft) HAAT with 100 kW ERP. Their site is 237 kM (148 miles) from KPEZ. In areas of weak RF, the less expensive receivers tended to jump to the adjacent channel. The only solution to this interference problem is more signal strength to the receivers from the Class A station.

In the midst of the strategy discussions, John W. Barger, Senior Vice-President of Clear Channel, asked if there is any way power can be increased to accomplish the goals of penetrating commercial buildings, fight adjacent channel interference and yet have height to overcome hills. This was answered by "Rule Making Process". Then the hours of testing combinations of power and height verses interference began with a goal of the most height with the most power.

INTERFERENCE STANDARDS

A standard was needed to make sense of the Separation Table of minimum distances between different classes. A study of the rules with calculations shows that the table is based on contours derived from the interference ratios of stations operating at the maximum power and maximum overall height.

The FCC Rules standard for allocations in the Educational FM Band is given in Para. 73.509 (b). The Desired (protected) contour is calculated with the F(50,50) curves. This is the 60 dBu (1 mV/m) contour for all stations. Class B stations presently received protection to the 54 dBu (0.5 mV/m) contour (not by rule, but by practice). The Undesired contour (interfering) is calculated using F(50,10) curves. The values of these Undesired contours were determined by the following ratios:

Co-Channel	40 dBu
1st Adjacent	54 dBu
2nd Adjacent	80 dBu
3rd Adjacent	100 dBu

A STUDY OF THE TABLE OF SEPARATION

The Separation Table and Class definition classes were changed as a result of Docket 80-90. A simple computer program was written to tabulate each channel showing the following data for each Class and each frequency position (Co-channel, 1st, 2nd, 3rd adjacencies):

Distance to the Desired contour
Distance to the Undesired contour
Distance of Separation from the table
Clearance of contours
(Desired+Undesired-Separation)

The results of the study showed that the Separation Table did a good job on Co-Channel, 1st adjacent and 3rd adjacent. However, for the most part, 2nd adjacent channels were consistently overlapped with the largest overlap of 36.5 kM (C to C).

NEW HEIGHT AND POWER

The program was expanded to include a new Class for comparison, Class *A. This was to be operated at more power and height than Class A with the results compared to existing clearances. A systematic method was employed starting with 5 kW ERP and 150 meters HAAT (500 feet). Then different powers and heights were reduced to find the optimum point of minimal overlap.

Consideration for what this would do to Co-Channel Class A stations was low on the consideration list. It was quickly determined that for Class A stations to take advantage of power increase, they would have to follow the example of Class IV AM stations who all increased their power and agreed to accept interference. If all stations increase the power uniformly, the point of interference between stations does not change. In directions where there is no close spaced station, there would be gains without interference.

Height Above Average Terrain of 125 meters and Effective Radiated Power of 4 kW was found to be ideal. The power gain is just under 4 dB when compared to the power reduction from 3 kW required at 125 meters. Protection results are as follows:

1. The Undesired Co-channel, 1st and 3rd adjacent channel contours either cleared or were tangent to the desired contours of all Class B and C stations.

2. The Desired 54 dBu contour of Class B was overlapped by the 1st adjacent Undesired 54 dBu of the Class *A by 3.2 Km but clears the Class B Desired 60 dBu.

3. The 2nd adjacent channel overlaps from Class *A were less than those given by all Class B and C stations.

The Class B stations were studied in detail. As was mentioned previously, Class B stations have protection to the 54 dBu based on the Table of Separation. Presently, the 2nd adjacent Class B located at minimum spacing overlaps the Desired 10.4 kM (but clears the 60 dBu). This translates into 305 square kM of area or 2.3 percent. A Class *A located at minimum spacing overlaps the Desired contour 3.2 kM. This translates into 82 square kM of area or 0.6 percent. Under Educational Rules, an overlap of interference of 1 percent is considered diminimus by the FCC. Figure-1 shows the graphical presentation. More information is given in later paragraphs.

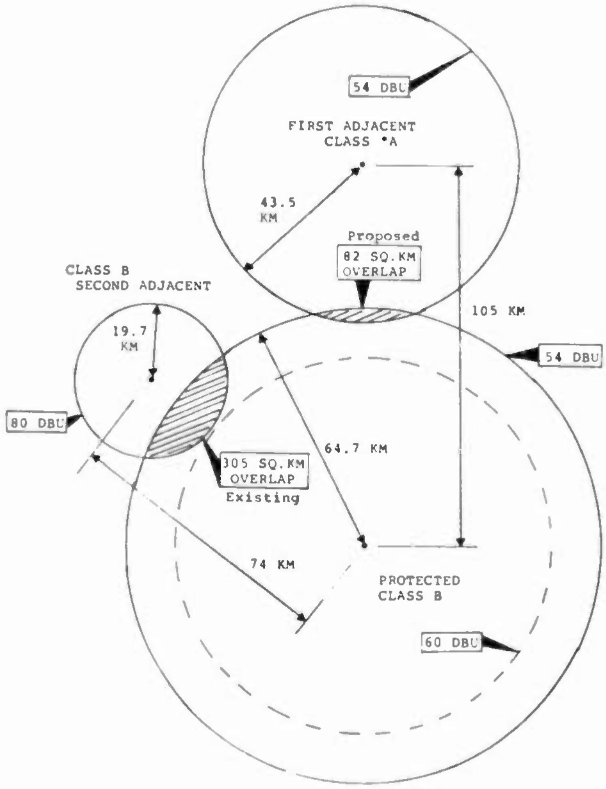
A CAUSE IN NEED OF A DOCKET

By the time the results were decided, the FCC had just started implementing the allocations that came from the action of Docket 80-90. It was felt that a petition might be considered as hindering this implementation, so it was postponed.

The Class A reserved channels limited to 3 kW has not been changed since the FM band was moved from 70 mHz in the 1930's. For the first time, the FCC has proposed changes in Docket 86-144. Part two of the docket proposed deletion of the special designation of Class A to the 20 reserved channels so any Class A station which could meet the separation requirements could upgrade without changing channels to one of the adjacent channels.

Clear Channel saw this as an opportunity to introduce this idea to help all Class A stations, not just a few that could up grade. The proposal was offered as "Comments" to this docket. On December 31, 1986 the FCC declined the Clear Channel Proposal but granted the proposal to delete the special designation of Class A to the 20 reserved channels. Clear Channel is planning more action at a future date.

FIGURE-1
CLASS B INTERFERENCE STUDY



NEW JERSEY ALTERS THE PROPOSAL

The New Jersey Class A Broadcasters Association, led by Bob McAllan of WJLK, filed comments modifying the Clear Channel Proposal. There are problems in New Jersey getting FAA approval for higher towers, so they requested modification to HAAT of 100 meters and ERP of 6 kW. This proposal was endorsed by Clear Channel in reply comments. The net effect on adjacent channels is the same. There is slightly more Undesired distance to Co-Channel Class A and conversion to 125 meters HAAT reduces power to 3.8 kW.

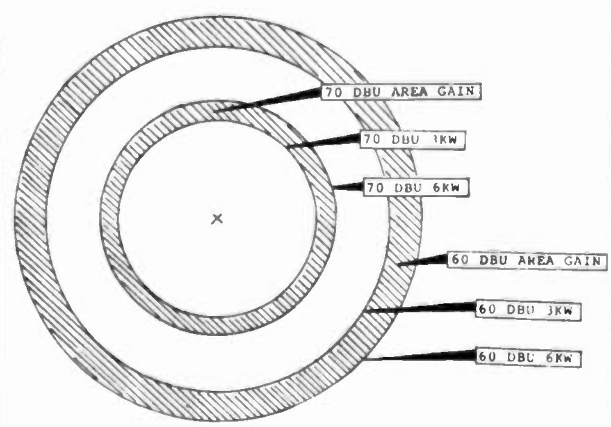
COVERAGE GAINS AT 6,000 WATTS, 100 METERS

A change from 3,000 watts to 6,000 watts is a 3 dB gain. Radius gains (moving a contour away from an antenna in a straight line) is a square root proportion to the power change. To double a radius distance, the power must be increased four times. A doubling of the power, then, is a 1.4 gain. Using the F(50,50), the 70 dBu (3.16 mV/m) contour increases from 13.6 kM to 16.1 kM. The 60 dBu (1 mV/m) increases from 23.8 kM to 28.1 kM.

A small radius gain with double power increase does provide opportunities that are indirect. Example: The next population center you would like to serve is 6 kM beyond your 60 dBu contour. Allocations permitting, a site could be located 2.5 kM closer to the new community and still maintain 70 dBu over the city of license. Also another 4 kM distance would be gained to the 60 dBu. Thus a net gain of 6 kM of distance is realized.

It would appear that a small radius gain would not result in much listener gain. Area gain with small radius changes can be quite significant. The area gained in the 70 dBu is 40 percent, and in the 60 dBu is 39 percent (see Figure-2). If the area is uniformly populated, then you have gained a listener potential increase of 40 percent in each contour. This gain is especially helpful if your station is located in an urbanized area where small gains can add thousands of new listeners.

FIGURE-2
CLASS A AREA GAIN WITH POWER INCREASE



PRACTICALITY OF POWER INCREASE

Adoption of this proposal will make power increase economically feasible for the low budget operations of many Class A stations. Many letters and phone calls came to Clear Channel after the Comments were filed with the FCC. An informal survey revealed that a large number of broadcasters already have transmitters that can be readjusted to reach 6kW ERP. Some will have to change the antennas to increase the gain. Depending on the antenna brand and the labor to install, this could cost from $5,000 to $10,000, usually with no other changes needed. A new transmitter capable of the required power will cost between $20,000 to $30,000 if that option is taken. Depending on the individual circumstances, a capital cost of $0 to $30,000 can be required, which is quite practical in terms of potential new revenue.

EFFECT OF DOCKET 86-144

The FCC proposed to help the Class A stations by removing the special designation of the 20 reserved Class A channels to allow a class jump without changing channels. A study was undertaken to see the net effect of this proposal. A computer study of some 2,500 licensees or allocations was made for all 50 states on all 20 reserved channels. In Region I and I-A, each Class A was studied as a Class B1. In Region II, each Class A was studied as a Class C2.

The basis for the study was made as follows:

1. If an assignment had an overlap of 20 kM or more, it was considered "Short" because the 70 dBu (city grade) of a B1/C2 is 20 kM and a site move would not allow the city of license to be served and therefore an upgrade was not possible.

2. If an assignment had an overlap of less than 20 kM and no other assignment in the opposite direction prevented a site move to clear short spacings, it was considered "Move".

3. If an assignment cleared all Co and Adjacent channels from the listed coordinates or had 1 overlap of not more than 1 kM, it was designated "Clear".
NOTE: The effect of a simultaneous move or upgrade without a move so the move or upgrade was mutually exclusive with another Class A, or whether the move would be able to cover the city of license was NOT considered.

The result of this study is shown in Figure-3. The effectiveness of the action of the FCC. must be considered with the following points:

1. Ten percent of the Class A stations can upgrade.

2. In Region I where the most dense population is located, about 2 to 3 percent will actually be able to upgrade to a higher class.

3. A capital outlay of $50,000 to $150,000 will be required for maximum service with an upgrade, or a site move. Land could cost more.

FIGURE-3
CLASS A UPGRADE STUDY

STATE	TOTAL	CLEAR	MOVE
AL	52	2	4
AK	17	11	3
AZ	40	12	10
AR	79	2	2
CA	166	19	23
CO	36	21	9
CT	10	0	0
DE	8	0	4
DC	0	0	0
FL	101	13	10
GA	82	3	2
HI	8	7	1
ID	21	8	6
IL	98	2	4
IN	75	0	1
IA	68	1	5
KS	38	1	2
KY	84	0	3
LA	54	1	3
ME	27	2	6
MD	21	0	0
MA	24	0	0
MI	97	8	12
MN	58	9	6
MS	66	2	1
MO	76	4	2
MT	19	14	3
NE	26	10	2
NV	14	8	2
NH	16	0	2
NJ	21	1	0
NM	31	7	13
NY	103	0	2
NC	50	1	5
ND	12	6	3
OH	81	1	3
OK	62	5	7
OR	26	11	3
PA	91	0	2
RI	3	0	0
SC	59	0	2
SD	13	5	0
TN	61	0	1
TX	204	13	19
UT	13	5	4
VT	20	0	2
VA	59	1	4
WA	33	5	7
WV	38	0	2
WI	75	2	2
WY	14	7	2
TOTAL	2,550	330	211
PERCENT	100%	13%	8%
ZONE I	867	17	46
I-A	166	19	23
ZONE II	1,517	194	142

4. A significant number of stations will be mutually exclusive in upgrade and legal costs must be considered.

THE CAPITAL BROADCASTING PROPOSAL

The "Capital Broadcasting-WLRX" and other Class A stations tendered Comments to the Docket 86-144 requesting that any Class A station should be allowed to increase its power based on the Educational Rules (outlined previously). This differs from the Clear Channel proposal by allowing power increases above 3 kW to the limit of the contour on an individual basis rather than a blanket increase for all Class A stations. As an alternative to the Clear Channel position, this proposal must be considered with the following points:

1. The ratios required by the Educational Rules are more restrictive on 2nd adjacent stations than the table of separations.

2. Very few of the Class A stations which cannot be changed to a Class B1/C2 will be allowed to upgrade because of the close spacing of co-channel stations and no allowance was proposed for acceptance of co-channel interference.

3. Almost every application to increase power will be mutually exclusive with other Class A stations seeking higher power.

4. A practical approach to minimize mutually exclusive applications would be to start in the center of the U.S. and work out in circles. Thus the outer circles would have the lowest power where the population is more concentrated.

A STUDY OF THE THEORETICAL INTERFERENCE TO CLASS B STATIONS WITH THE CLEAR CHANNEL PROPOSAL

It has been stated previously that the theoretical overlap from a Class *A to a Class B 54 dBu contour is 0.6 percent, a value that the FCC has already defined as diminimus. If the FCC adopts the Clear Channel proposal allowing Class A stations to increase ERP to 6 kW, how many actual Class B stations would actually receive interference within the Desired contour? All 761 Class B assignments or licenses have been studied and the effect is reported here.

The basis for the study was made as follows:

1. The Table of Separation was altered to provide 108 kM separation for 1st adjacent Class A to Class B. This number was derived from existing separation of 105 kM plus 3 kM overlap of the Undesired 54 dBu.

2. If no Class A stations were short spaced then the Class B station was designated "Clear".

3. If a Class A station was the only short space or the greatest short space, then the Class B was designated "1st SHORT".

4. If a Class A station was short spaced, but the Class B has existing short spacings by other class B/C greater than the short spacing of the Class A, then the station was designated "SHORT B>A".

NOTE: Because of the large number of "SHORT B>A" stations and a limit of study time, an examination that all of the short spacings of the Class B's were in the same general direction of the Class A's was NOT done. However, a spot check of data showed that there was consistently a short spaced Class B in the general direction of the offending Class A.

The result of this study is shown in Figure-4. There were 761 assignments studied in 19 states for 1st Adjacent Class *A to Class B using a new separation of 108 kM. A summary of the data is as follows:

FIGURE-4
CLASS B INTERFERENCE STUDY

STATE	TOTAL B's	1ST SHORT BY DIST.	1ST SHORT BY CONT.	SHORT A<B
CA	162	9	7	10
CT	14	3	2	6
DE	3	0	0	1
DC	8	0	0	1
IL	63	6	2	9
IN	46	3	2	11
ME	23	1	0	1
MD	28	1	1	9
MA	32	2	2	7
MI	51	3	0	13
NH	6	1	0	1
NJ	19	0	0	6
NY	70	4	1	11
OH	90	1	1	21
PA	76	6	5	27
RI	6	0	0	2
VT	3	1	0	0
VA	34	1	0	2
WV	27	4	2	1
TOTAL	761	45	25	139
PERCENT	100%	6%	3%	18%

"SHORT B>A" Class B has existing short spacings greater than Class *A.
"1st SHORT-DIST." Site short space less than 108 kM, first new short-space.
"1st SHORT-CONT." From those 1st Short-Dist. whose contours actually overlap based on terrain.

A STUDY OF THE ACTUAL INTERFERENCE TO CLASS B STATIONS WITH THE CLEAR CHANNEL PROPOSAL

Each station in the "lst SHORT" category was studied in detail to determine the amount of actual overlap based on predicted contours. The following basis of the study was used:

1. A Terrain Profile computer program using the NGDC 30-second data base to determine the HAAT of each radial of the Class B and Class A facility.

2. The distances to the Desired 54 dBu were calculated using a computer program with the most recent metric curves. The area of the Class B was calculated by formula using the distances to the 8 radials. NOTE: The sites were not plotted on maps to have areas in foreign territory or large bodies of water removed.

3. The Class A's Undesired contours were also calculated using the same basis as (2) above.

4. The pertinent portions of the contours were plotted on a simple scale form of 1:1,000,000 in lieu of a map. The areas of overlap were measured by use of a compensating polar planimeter.

5. Areas of existing overlap from Class A at 3 kW were subtracted from the overall area and new overlap area, where they occurred.

Figure-5 shows the call signs and city of license of the 25 stations which would receive interference. The call sign of the offending station, the distance of separation, the maximum radial overlap and the area of overlap are shown. A ratio of present area compared to overlap areas is shown in a percentage. A sum of all the areas with a sum of all the overlap areas reveals that there would be 1.54 percent overlap.

CONCLUSIONS

1. The goals stated in the abstract can be reached by increasing the power of Class A stations. This has been shown by field measurements and other tests.

2. The FCC's action of dropping the special reservation on the 20 channels will benefit about 10 percent of the Class A stations with significant capital and potentially other legal costs.

3. The "Capital Broadcasting" proposal to raise power has weaknesses as relates to Second Adjacent channels and possibly every application would be mutually exclusive.

4. The Clear Channel proposal helps all 2,550 Class A stations increase their coverage 40 percent with diminimus of interference and minimum of capital and legal expenses.

FIGURE-5
SUMMARY OF OVERLAP DATA FOR THE 3% OF CLASS-B STATIONS WITH 1ST INTERFERENCE FROM CLASS-A

B-CALL	CITY,STATE A-CALL	DIST. Km	OVERLAP Km	TOTAL AREA AREA Sq. Km	54dBu PER-CENT
KDDB	Paso Robles, CA			12,019	
	KBOX	105	-4	75	0.6%
KAMB	Merced, CA			11,914	
	NEW	106	-4	103	0.8%
KSJO	San Jose, CA			13,413	
	KINQ	71	-6	612	4.6%
KCBQ	San Diego, CA			12,496	
	KHYE	104	-8	312	2.5%
KKHI	San Francisco, CA			13,508	
	NEW	108	-2	20	0.1%
KMYT	Merced, CA			12,899	
	KWIN	102	-7	176	1.3%
KNTI	Lakeport, CA			13,500	
	KVYN	102	-7	148	1.1%
WRKI	Brookfield, CT			12,399	
	WSBH	107	-6	150	1.2%
WTIC	Hartford, CT			12,547	
	WYRS	101	-5	120	0.9%
	NEW	105	-8	204	1.5%
WEBQ	Harrisberg, IL			13,061	
	WJBDFM	106	-4	71	0.5%
WSMI	Litchfield, IL			12,574	
	WSAK	104	-4	94	0.7%
WGMI	Terre Haute, IN			13,082	
	WBCI	105	-4	95	0.7%
WFIU	Bloomington, IN			12,932	
	WRBI	107	-6	151	1.2%
WFRB	Frostberg, MD			13,006	
	WWOO	106	-8	387	3.0%
WROR	Boston, MA			11,990	
	WILIFM	107	-6	133	1.1%
WBEN	Buffalo, NY			27,566	
	WFLC	104	-9	479	1.7%
WBNS	Columbus, OH			13,229	
	NEW	105	-7	135	1.0%
WMYG	Braddock, PA			13,145	
	WKTX	104	-5	90	0.7%
WJNL	Johnstown, PA			12,645	
	WQWK	101	-9	478	3.9%
WIYQ	Ebensburg, PA			12,757	
	WPQRFM	103	-6	204	1.6%
WHTX	Pittsburgh, PA			12,984	
	WMGZFM	104	-7	279	2.1%
WCCK	Erie, PA			14,418	
	WIFI	105	-8	320	2.2%
WVAQ	Morgantown, WV			12,787	
	WRRRFM	107	-3	39	0.3%
WQBE	Charlestown, WV			13,057	
	WCJO	103	-5	129	1.0%

TOTAL AREA CLASS-B: 323,928 SQ.KM.
CLASS-A OVERLAP AREA: 5,004 1.54%

TIME DELAY AND PHASE ERROR DETECTION AND CORRECTION

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Presenting for
Howe Technologies Corp.

I. Abstract

Parallel channel analog recording and transmission systems are susceptible to interchannel time delay errors which give rise to anomalies in stereo image localization and cause frequency response degradation in monaural reproduction. The various causes of differential time delay errors are identified, and the subjective effects are discussed. Two measurement and correction techniques are outlined, and recommendations concerning applications and procedures are proposed.

II. Introduction

With the introduction of multitrack analog recording, inter-channel time delay error became a production reality, giving rise to a variety of unwanted auditory artifacts and distortions, including poor stereo imaging and degraded monaural frequency response. The widespread use of the stereo tape cartridge, particularly in broadcast applications, brought the problem into the spotlight. The mechanical compromises required to make the tape cart a cost effective medium left a great deal to be desired in the tape handling department, and as a result, even precisely calibrated tape heads did not guarantee accurate alignment from cart to cart. The result was varying azimuth errors, which degraded stereo phase tracking and produced mono incompatibilty.

While the audio consumer is developing increasingly discriminating taste, with the digital compact disc, hi-fi VCR, and multi-channel cinema providing excellent standards of comparison by which other types of audio are judged, most radio and TV receivers, home video cassette machines and movie theatres are still mono. As high quality stereo programming proliferates in a world populated with mono machines, the problem of mono compatibilty becomes increasingly serious.

Ultimately, mono compatibility must be the responsibilty of the program producers and broadcasters. To guarantee that stereo audio will be mono compatible, accurate measurement of potential error conditions must precede any attempt at correction. To do this, we must first develop an adequate definition of the quantity to be measured before constructing the measurement apparatus itself. Because of the subtle nature of the time and phase delay errors and the often confusing terminolgy found in current literature, it seems prudent to clarify several terms at the outset.

III. Definition of Terms

A sinusoidal signal has three degrees of freedom: amplitude, frequency and phase. Throughout this paper, only amplitude and phase will be used as independent variables. Differential time delay error refers to an unintentional time shift between two related channels; similarly, differential amplitude error is an unintentional inter-channel level imbalance.

It should be pointed out here that phase shift and phase delay are used interchangably to mean linear phase shift as a function of frequency, that is to say, to pure time delay. The type of phase shift caused by analog frequency shaping filters is not generally linear with frequency, and the original phase relationships between signals of various frequencies are not preserved. Though not strictly a time delay problem, it is often a useful concept and will be referred to as phase dispersion.

An important distinction that is too often overlooked is the difference between a 180 degree phase shift at a particular frequency and a polarity reversal. For a symmetric, single frequency waveform, such as a sine or triangle wave, a time delay of 1/2 period (cycle) is equivalent to a phase reversal. However, for non-symmetric or harmonically complex material these two conditions are NOT equivalent. A polarity reversal refers only to an actual inversion of the waveform. Therefore, it is impossible (except in the special case of a symmetric waveform) to correct a polarity reversal by introducing any amount of time or phase delay.

Conforming to popular usage, monophonic (mono) and stereophonic (stereo) refer to one and two channel audio program material, respectively. The term binaural will be reserved for the experience provided by a pair of headphones; this, incidentally, represents the ultimate near-field environment. The salient characteristic of a binaural environment is that there is no acoustic coupling between ears, so the precise nature of the material presented to each ear can be controlled independently.

For completeness, sound presented to only one ear at a time is considered monaural (as distinct from monophonic or "mono"); this case is of clinical interest, but will not be dealt with here.

When viewed as an acoustic information processing system, a person may be considered to be a two channel receiver provided with two spatially distinct, but otherwise similar, transducers. Each channel features a bandwidth approaching 20 kHz and a dynamic range exceeding 110 dB. Besides the two input channels, people also seem to have a built-in neurological stereo correlation processor which is especially sensitive to differences in inter-ear arrival time (for similar signals). The small inter-ear phase differences resulting from slightly different audio path lengths (equivalent to auditory parallax) are used as primary sound localization cues. As convenient bench marks, a sound wave travels 1 cm in about 30 usec, that is, 331 m/sec. Therefore the wavelength of a 1 kHz sine wave is 33.1 cm, so a path difference of about 1 cm is equivalent to a 10 degree phase shift.

Distinctly perceptible image shifts are produced by inter-ear arrival time differnces on the order of fifty to a few hundred microseconds; the ability to perceive such phase differences as localization cues is particular acute in the lower midrange, within a octave or so of 1 kHz.

Spatial localization of a sound source can be achieved either by utilizing inter-ear phase differences or relative amplitude levels, provided that some degree of correlation exists between the signals. If the degree of correlation is increased to unity so that both channels are identical, we have the effect of mono, leading to the the familiar phantom center effect in a two-speaker environment. Conversely, as the degree of correlation between the two channels is reduced to zero, the impression of a stereo image "spread out across the stage" vanishes, and is replaced by the impression of "two sounds from two speakers."

Because differential amplitude levels are easy to obtain in practice and serve as effective position cues, the stereo pan control (amplitude balance) on mixing desks has become virtually ubiquitous, whereas stereo panning based on parametric control of inter-channel time delay is rarely used.

IV. The Symptoms of Spurious Time Delay

A differential inter-channel time delay error is a meaningful concept only if two a priori conditions are met. The first is that there are two simultaneous active channels, and the second is that there is some degree of simularity between them.

For a stereo source, a differential time error results in a simple rotation of the stereo image away from the delayed side, corresponding to an increase in virtual path length. If the sound source is two track mono, the single virtual sound source shifts in apparent position.

In practice, time delay errors less than 50 microseconds produce almost insignificant imaging errors when perceived in a normal listening situation; when heard binaurally, the image shift becomes detectable at somewhat smaller time delays. Subjectively, the image disruption does not become objectionable until the delays exceed 100 microseconds or so, unless the listener is specifically made aware of the effect.

Time delay errors have more serious consequences for the monophonic situation, as, for example, when a stereo broadcast is heard on a mono receiver. If a time delay T is introduced between two identical signals which are then mixed together, it produces the classic comb filter response, which is a series of amplitude nulls in the frequency response, spaced evenly at harmonic intervals.

The first amplitude null in the output occurs at a frequency having a period equal to twice the time delay. This is because all signals of that frequency will experience an inter-channel phase shift of exactly 1/2 period, or 180 degrees, causing them to cancel when the two misaligned channels are mixed together.

Frequencies below the first null will exhibit attenuation similar to the roll-off effect of a standard treble control; above this first notch, additional nulls will occur at every harmonic of the first null, leading to the familiar "phasing" effect.

In general, for a given δT, the relative amplitude of a particular frequency (f) is proportional to $\cos(\pi f \delta T)$.

To put this analysis in perspective, a real-world example is useful. Consider a 1/4" two track tape player which runs at 7.5 ips, with a nominal spacing between the two linear audio tracks of about 1/8". If the playback head is slightly out of alignment, say 0.5 degree, the effect is equivalent to longitudinally offsetting the two tracks on the tape by 0.0011". At 7.5 ips, this translates into a differential time delay error of 145 microseconds, which produces a complete null at 3.45 kHz and 50% power at 1.72 kHz. Clearly this sort of high frequency loss is something to be avoided.

V. Sources of Time Delay Error

Any time two originally synchronized signals are transmitted through physically distinct channels, differential time delays can occur. The most well known example of this is due to head azimuth misalignment on linear track analog recording systems, including both magnetic and optical types (see Note 1). The magnitude of the resulting time delay is related to track separation distance, tape or film speed and transducer alignment angular error.Another more subtle source of error is found in multichannel analog recorders. A critical characteristic of every mulitchannel head stack is the precise vertical colinearity of the pole piece gaps; any deviation from precise alignment is termed gap scatter. As long as a tape is played back on the same machine on which it was recorded, gap scatter does not produce a problem.

Switching machines between recording and playback can produce inter-channel time delay due to slight mismatches in pole piece alignment in the different heads. Fortunately, using more than two simultaneous tracks (i.e, stereo) for one source is rare, so a single pair-wise time base correction is all that would be required.

Discrete multi-channel transmission links are susceptible to inter-channel time delays. For example, in a discrete STL microwave application, slightly different processing electronics or transmission path lengths could produce sufficient time delay to cause mono compatibility problems later. Likewise, repeater electronics in satellite systems can introduce time delay errors. Interestingly enough, multipath reception can also become a problem in studio situations where two widely separated microphones "hear" the same signal; a few additional feet in one of the paths can produce some interesting effects during mixdown (such spurious effects can be detected and corrected using the appropriate equipment).

The latest addition to the list of time delay error generators is found in compact disc players. Some systems use only one D/A converter for both channels, which produces a half sample period delay (11.34 usec) between the channels. This represents a 59 degree phase error at 15 kHz, which causes an error of 1.21 dB in the mono sum.

VI. Time Delay Error Detection and Correction

There are two possible approaches to time delay error detection and correction, corresponding to the explicit or implicit nature of the information which is common to both channels. If the form of the common information is known, then standard correlation techniques can be employed to find the differential time delay. Two detection schemes, the a priori and post hoc methods, respectively, will be described and then the benefits and drawbacks of each will be discussed.

In the a priori system, a characteristic pilot tone constitutes the "known common information" and is added simultaneously to both audio channels during recording. Upon playback, the pilot tones from each channel are cross-correlated to produce an error term proportional to the differential time delay. This error signal is used to control a pair of time delay circuits which adds an appropriate amount of delay so that the pilot tones match, implying that the proper time base relationship has been restored.

In a one currently available system, a 19 kHz pilot tone, amplitude modulated at 300 Hz, is used as the encoding signal. Upon playback, the pilot tone on each channel is demodulated to extract the 300 Hz signals, which are correlated to get a first approximation for the time base error. With the error determined by the first correlation within one cycle of the 19 kHz tone, a second phase comparison at the high frequency establishes the precise time base error. This information is used to control the output clock timing of a pair of digital delay lines which provide the required time delay correction.

The *post hoc* approach differs from the first scheme in that no prior encoding of the program material is required; all necessary information needed to establish proper time alignment is derived from the program material itself.

At this point, it must be recalled that normal stereophonic material is not merely two simultaneous audio programs, but is in fact a highly correlated pair of audio channels. In addition, it is assumed that the average stereo image is intended to be symmetrical about "center stage." If these two assumptions are valid, then a cross-correlation between the actual audio channels will provide accurate time base skew information. Fortunately, it turns out that this method yields exceptionally good results in practice, and is therefore the basis for the post hoc system.

Two points will need further clarification. The first is, precisely what audio signal characteristics are used to determine the original interchannel time relationship, and second, how can the intentional phase variations which characterize stereo material be reliably distinguished from the phase errors caused by the unwanted time delay?

For the range of program material commonly encountered, including radio music programming, dialog, and the complex sound tracks of stereo television and feature films, it turns out that the frequency band between 800 Hz and 16 kHz contains enough information to characterize the interchannel time delay.

Bandwidth limited signals from each input channel are first processed by AGC amplifiers to eliminate any possible errors due to amplitude variations. These normalized signals are then cross-correlated to yield an output proportional to the instantaneous differential phase; this output represents the sum of normal stereo phase fluctuations and phase differences due to time delays.

Normal stereo phase fluctuations are quite rapid, while fixed time delays produce essentially steady state values. Since it has been assumed that the average phase deviation of proper stereo is zero, a good measure of time delay error can be obtained by integrating the cross-correlator output signal: any deviation from zero is proportional to time delay error.

A complete audio time base corrector is obtained when the integrated output of the cross-correlator is used to drive a DC servo amplifier connected to a pair of time delay circuits as shown in Figure 1. Such a closed-loop negative feedback system will continuously provide just enough time delay correction so that the average phase error between the audio output signals is zero.

By correctly tailoring the feedback loop characteristics, the residual error of the time base corrector can be made quite small; for sine wave signals, the correction error typically is less than 1 microsecond.

In order to avoid providing correction where none is required, a "smart" stereo image processor circuit is needed. It must ignore "artistic phase variations," yet be able to accurately respond to large changes in differential time delay that often occur between program segments.

These requirements are fulfilled by incorporating the electronic equivalent of a "window of zero correction" into the system servo loop. It is implemented as a dual time constant integrator with a window detector in one side chain (see Figure 1).

When a time delay error is introduced that is large enough to cause the output of the fast integrator to exceed the limits of the "window", a high speed correction circuit takes over and rapidly changes the time delay correction to "center" the stereo image. A proprietary circuit design allows this step detector to operate very rapidly with almost no overshoot; thus abrupt time delay error changes can be corrected without severely compromising normal stereo material.

For a properly centered stereo signal, both the slow and fast integrators will have a dc output of zero, corresponding to a system state of "no net correction." The slow, long time constant integrator effectively ignores all the normal phase fluctuations of the stereo signal, so no erroneous correction is applied to normal signals.

The feedback loop topology provides an additional feature at no extra cost. The differential time delay networks have a linear transfer characteristic, so the control voltage required to balance the circuit is directly proportional to the input time delay error. By using this control voltage to drive a front panel display, an accurate real-time indication of input phase condition can be provided.

In addition, when the cross-correlator output, prior to the integrator, is displayed on a standard oscilloscope, it provides an extremely accurate representation of instantaneous inter-channel phase differences and is useful both as a diagnostic tool and as a precision calibration reference.

VII. Discussion

It is clear from the preceding discussion that both of the possible schemes for audio time delay correction are technologically feasible and potentially useful. The question now becomes one of application.

Because the a priori method described first relies only on the accuracy of the original pilot tone and the phase detectors in the decoder processor, it has the potential of achieving almost perfect audio time base correction both for short term errors such as wow and flutter as well as for long term differential time delays. There are however two drawbacks to its universal application.

The first drawback is that this kind of system will correct time base errors only for material that has been processed to include the pilot tone on each audio track. Time delays produced by microwave links and discrete telephone lines, for example, cannot be corrected by an encoder-decoder process unless the material is encoded at the source, which is usually impractical.

Secondly, if the source material is already recorded in stereo, perhaps with unknown intrinsic time delay errors already present, then re-recording the material to include pilot tones would have no advantage. On the other hand, if one is in the process of creating an in-house tape cart library from high quality original source material (such as phonograph records), the benefits of such a pilot tone system are obvious.

The *post hoc* system has its shortcomings as well. The most often asked question is, what will it do to stereo material which incorporates intentionally phase shifted material? The answer, unfortunately, is simple: if the phase deviation is big enough for long enough, the system will eventually center the image. On the bright side, this unwanted correction effect occurs rarely if ever in practice, since a centered stereo image is the general rule.

The other problematic situation concerns system operation when presented with complex stereo material produced from several different source tapes. This occurs quite often with stereo programming produced for television and feature films. Typically, the music, dialog and effects tracks came from different sources, but have been mixed down to make the final two track master. Field data indicate that occasionally the original sources must have contained time delay errors, which are individually measurable in the final mix. The important question here is, what happens when the mix contains material with different intrinsic delay errors?

The *post hoc* system derives its information from the program material itself, and ultimately behaves as an average responding device. Therefore, if there are several different "phase errors" occuring simultaneously in the program material, the system will provide correction corresponding to the algebreic averaqge of the individual phase errors.

Once again, actual field data indicate that the occurance of equal levels of "phase conflicting program material" is not common. Instead, it is usually the case that when dialog is present, the music fades, and when the music comes up, no one is speaking. The correction circuitry dutifully follows the predominant material, applying the appropriate time base correction (see Note 2).

Usually the foreground audio contains the most relevant material, so the "cocktail party effect" takes over: people pay attention to what is important at the time, based on semantic considerations. This tends to provide perceptual masking for other backgound information.

As mentioned previously, the system is designed to rapidly adjust the time delay correction, and it tracks the foreground material. The background material, although perhaps not corrected properly (particularly if it has different phase problems than the foreground), is largely ignored. The net result, however, is that the lack of perfect correction is irrelevant to the listener in most cases.

Clearly this is a generalization, but it demonstrates the underlying design premise: when correcting multiple errors, strive for the most artistically coherent compromise. The assumptions that have been made in the design of the various decision circuits stand up quite well in actual listening tests and seem to produce acceptable results even under adverse conditions.

VIII. Recommendations

There are a number of situations where proper use of audio time base correction systems can yield significant operational benefits. However, it is not recommended that they be used as universal replacements for regular maintenance and good engineering practice.

For on-line broadcast applications, the inclusion of an automatic correction system in the audio chain serves as an insurance policy, guarding against poorly mastered source material and various types of equipment failure, such as cart machine head failure. The visual display of input time delay status allows control room personnel to see errors as they occur. Such real-time monitoring capability of inter-channel phase significantly augments the information derived from a standard vectorscope display.

In highly critical environments such as the master control of a large facility, it is recommended that a real-time phase monitoring device be used at all times; if possible both a standard vectorscope and a real-time linear phase display should be available. If errors are detected, then every effort should be made to correct them before show time. In the unfortunate event that satisfactory corrections cannot be made in time, the correction device could be patched into the line as a last ditch solution.

Once a multitrack master tape is mixed down, any time base errors present in the source material will become convoluted and hence fundamentally uncorrectable. It is strongly recommended that a means of accurate time delay error monitoring be included in all master recording and post-production studio designs. Additionally, when accurately recorded source material is unavailable, the use of audio time base correction equipment is recommended to correct stereo time delay errors prior to final mixdown (see Note 3).

The ability to detect and automatically correct audio time base errors without operator intervention has obvious applications in automated tape duplication operations. By eliminating time base errors directly in the program line before duplication, costly retakes are avoided and customer satisfaction is increased.

IX. Conclusions

The weakest link in a process determines the quality of the product, so the negative impact produced by mono incompatibilty of stereo material should not be underestimated. All too often a simple inter-channel time delay error is the only problem in an otherwise superior product.

With the availability of precision time delay measuring and correcting equipment, producers, sound engineers and broadcast personnel can now begin to recognize and quantify the sources of time delay error and develop analytic methods to eliminate these problems. Therefore, the use of a laboratory grade instrument which provides both the standard vector display of the stereo channels plus a simultaneous linear display of instantaneous inter-channel phase difference is recommended as a diagnostic tool.

Clearly all sterophonic material may not be completely compatible in a mono mix. However, if the nature of the various mix-down compromises are known at the beginning, then informed choices can be made between "good stereo" and mono compatibilty.

X. Notes

Note 1.

Recent field tests have shown that stereo optical transducer systems are also subject to severe alignment errors, corresponding to head azimuth errors in magnetic recorders. The required alignment procedure is difficult and the results obtainable by highly skilled field technicians is often less than perfect.

Using the standard production unit, the calibration procedure was greatly facilitated and the results obtained were equal to or better than any previous results. Furthermore, employing the device in the audio chain following calibration removed all residual time delay errors attributable to the mechanical system, as well as correcting time delay errors detected in the test film itself.

Note 2.

A variety of program material has been examined in our laboratory and in the field; preliminary data indicates that the vast majority of otherwise high quality stereo program material suffers from various amounts of time delay errors.

The range of material tested included music videos (broadcast programs and original production masters tested at the post production facility), prime time network stereo broadcasts and original source tapes tested on-site, live stereo television audio production (monitored on-site during taping), FM and AM radio broadcast programs (including music, talk shows and syndicated special programs), and optically recorded feature film Dolby audio (tested on site).

In every case, the real-time unit was able to detect and correct differential time base errors, including composite errors contained in multi-track material.

Note 3.

This situation occurs principally in the original multitrack mastering environment. If a time delay monitoring device provided with a linear display of instantaneous phase error is used as a monitor on the final stereo mix, occasional program phase anomalies can be easily spotted.

Because of the effectively "microscopic" resolution of the novel CRT display, the engineer can, upon replay of the mix, readily identify which of the stereo input sources contains the time delay error. This error can then be corrected, allowing the mixing session to procede.

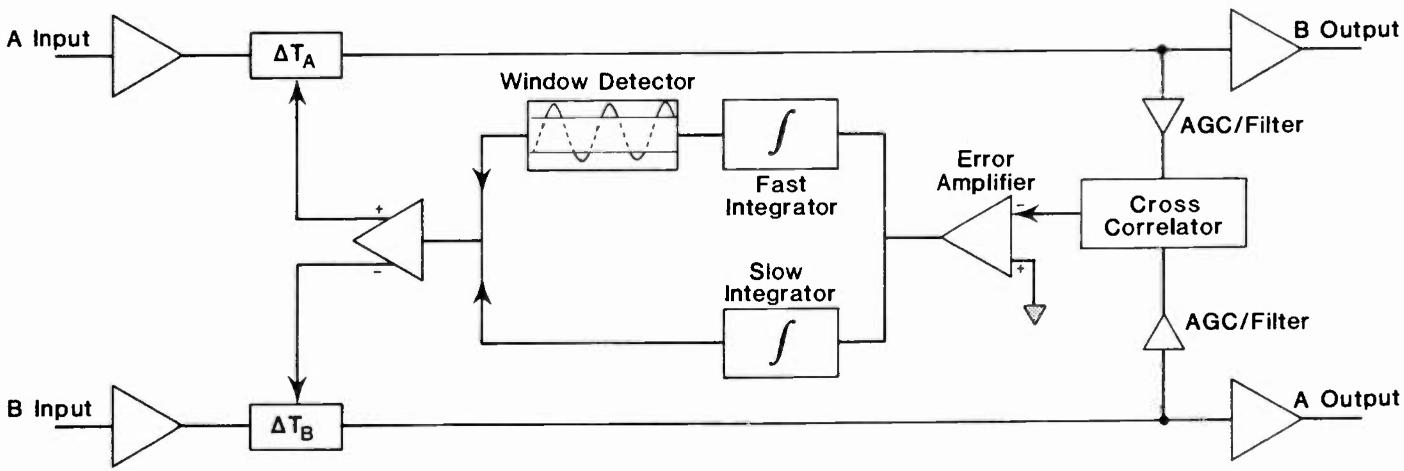


Fig. 1. Audio Time Base Corrector.

LIVE, VIA-SATELLITE RADIO REMOTES FOR THE WALT DISNEY WORLD 15TH ANNIVERSARY PARTY

Bruce Bryson
Walt Disney World Company

Ralph Beaver
WRBQ Radio
Tampa, Florida

INTRODUCTION

On October 1, 1986, Walt Disney World celebrated it's 15th Anniversary; 15 years of fun and magic. Birthday invitations were sent to media representatives across the nation and around the world. Many radio stations sent their shows to Central Florida to participate in this media event. This paper reviews the enormous complexity and technical challenges met by Walt Disney World cast members.

CAST OF CHARACTERS

Bruce Bryson

A Walt Disney World cast member since 1978, Bruce's formal education and background are in digital electronics and has worked with Epcot Computer Central Systems, the nerve center of Epcot Center. Bruce now coordinates satellite transmissions for "Worldlink", the satellite operations department of Walt Disney World, and is responsible for live broadcasts and taped transmissions. During the 15th Anniversary press party Bruce was responsible for technical coordination of all radio remote broadcasts.

Ralph Beaver

Beaver, who has been hired by Walt Disney World as a communications consultant in the past, was brought in to help with coordination of this event. Has been associated with radio satellite transmissions since the opening of Epcot Center in 1982. He enjoys the opportunity to meet fellow broadcasters from around the country and compare notes. He has been with Edens Broadcasting since 1969 and presently serves as director of legal and technical operations for WRBQ AM/FM..."Q105" in Tampa.

WALT DISNEY WORLD RADIO REMOTES

Walt Disney World is "wired for sound" with fiber optics and other special circuits networked throughout the property for remote radio and television broadcasting. This futuristic system has redundant stereo capability which provides reliable, top quality audio paths to the Worldlink technical center. The uplink is equipped with a digital encoder for stereo 15k, which makes it one of only three sites in the country capable of uplinking digital audio, and multiple SCPC (Single Channel Per Carrier) modulators for stations requiring analog service. These capabilities offer local radio stations network quality which when combined with the unique nature of the Walt Disney World Resort, has helped make it one of the most popular radio remote destinations in the world.

In addition to many regularly scheduled radio remotes, Walt Disney World hosts major media events which involve multiple simultaneous radio broadcasts. The first of these was the 1982 opening of Epcot Center which introduced Disney to utilizing satellite TV technology for radio.

The opening of The Living Seas pavilion in 1986 gave Disney an opportunity to get it's feet wet with underwater remotes live-via-satellite. This 6 million gallon salt-water aquarium in which Epcot Center guests go "underwater" was extremely successful. It also created the opportunity to handle a lot of radio stations simultaneously using dedicated radio uplinks rather than audio sub-carriers on full video uplinks.

EACH REMOTE BROADCAST POSITION

Each broadcast position at the Magic Kingdom and Epcot Center were provided a custom painted sign with the station's call letters, city and courtesy airline mention, chairs, a decorated table, and a lamp for early morning and evening remotes. The positions were also supplied with electrical and telephone drops, voice couplers, and telephone instruments in addition to a dry pair for satellite positions. Since each location was created for this event, all special wiring and equipment had to appear and disappear during the week to maintain Disney's reputation for "good show quality". These processes were repeated many times to accommodate the 260 radio remotes which aired during the event.

AUDIO TECHNOLOGY

Dry Pair

"Dry pair" refers to two wires from one point to another with no other telephone company equipment connected to them. This type of circuit is normal for most remote broadcast audio uses as long as they are less than 2000 feet, the audio level is kept around the standard "+4 or +8", and the impedance is kept lower than 1000 ohms. All radio stations were using equipment within these ranges. When radio broadcast locations on the Walt Disney World property were greater than 2000 feet from the radio production facilities, other types of specially treated circuits were used.

Remote position identifiers

In order to be certain the dry pair was good from the table to the Walt Disney World satellite operations center, each broadcast position had an individual battery-powered tone burst oscillator that kept a constant identifier on the circuit. Once this "first mile" was confirmed, the tone burst was then fed to the satellite channel so the receiving station would be assured of a signal from their broadcast position in Florida to their studio master control. This end-to-end testing was done to verify that all circuits would be operating when the stations needed them, since trouble-shooting time would be reduced to precious seconds should there be a problem .

Production facility

The two radio production facilities were located four miles apart at the Magic Kingdom and Epcot Center, and were responsible for monitoring and conditioning the audio feed from each remote broadcast position. These signals were then combined and fed to the uplinks. This facility was also responsible for monitoring the downlink to assist each station in fine tuning their live program feed.

IDB Communications, a supplier of satellite transmission and distribution services for the radio broadcasting industry. Gilbert Kuang and Howard Miller, normally in charge of IDB's New York and Los Angeles technical operations, were each given the task of program distribution from the Magic Kingdom and Epcot Center. Each center was staffed by a dedicated team of broadcast oriented personnel flown in from all over the country.

Uplinks

Three transportable uplinks were used for the radio transmissions, along with eight uplinks needed for the television portion of the 15th Anniversary festival. Television-style transmitters were used to provide the power necessary for the large quantity of audio carriers per transponder. One of the uplinks at Epcot Center transmitted directly to T15, the emergency digital network transponder on Satcom 1R, which was the first time radio stations across the country had an opportunity to test their standby crystal. This also was the first time local radio stations could use single-hop digital audio for their personal remotes.

Why use digital channels

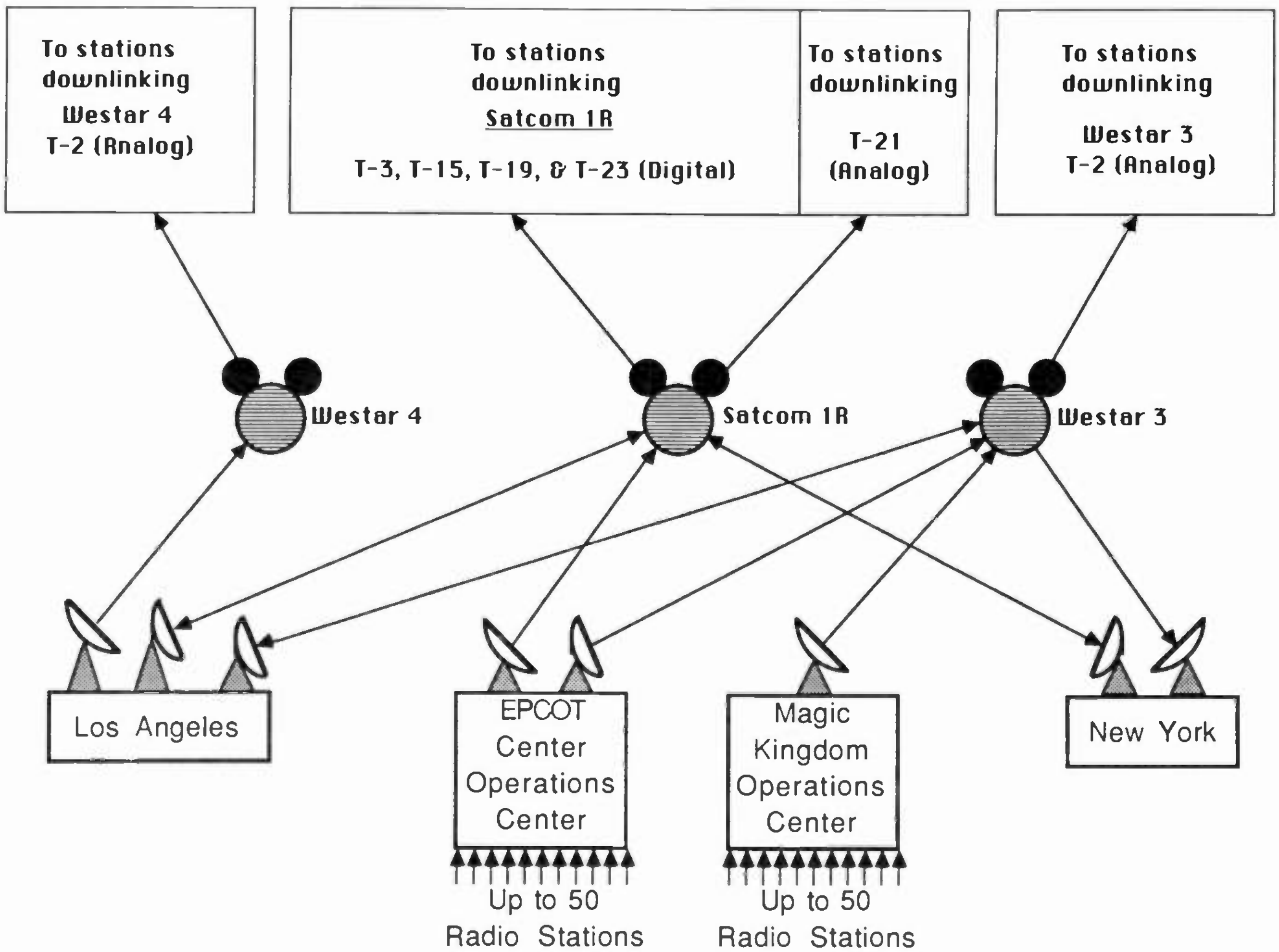
Any radio station receiving a major radio network via digital audio receiver only had to tune their unit to another channel to receive their personal signal from Walt Disney World.

Why use analog channels

Since all digital channels were full, SCPC analog channels served stations with the ability to receive them. These single-hop signals were also preferred by radio stations engaging in a talk show style format with live telephone calls.

New York and Los Angeles switching

To send the audio to network affiliates on Satcom 1R transponders 19 and 23, signals were received through IDB's New York facilities and routed to the major radio networks uplink. The signals take this path because these full-transponder digital channels originate from New York.



Radio Satellite Distribution for The Walt Disney World 15th Anniversary Celebration

Similarly, audio sent to stations via Satcom 1R transponder 3 were turned around through IDB's Los Angeles facilities. The signals take this path because these T3 digital channels originate from Los Angeles.

Radio use of video subcarriers

Sometimes a buzz is experienced in the audio signal of a video program. While this is sometimes tolerated in TV, it is not acceptable in broadcast radio. Picture buzz in the audio could be mis-adjusted sub-carrier or the result of a small receiving dish. In radio, the average size receiving dish is 3 meters or less, but in TV the average size is 5 meters or more. In any event, this is not the best method to deliver audio signals to broadcast radio stations. The number of available audio channels on any one TV transponder is also limited. Radio people go for hours at a time, but TV uses drop-ins of 5 minutes average length. The cost of space segment time varies greatly since one TV channel occupies one satellite transponder, and many audio carriers can be placed in one transponder.

Pitfalls of using telephone lines

While the dial-up telephone line is used for news and other on-the-scene reporting, this method of audio signal transmission is not normally tolerated by the radio listening audience for longer programs. When a broadcast station does a full-length remote program, a higher quality signal path is needed. Even the "not by satellite" radio broadcasts used some form of frequency extender to improve the standard telephone circuit sound.

THE LEARNING PROCESS

Satellite channels and formats

During this event, all three radio broadcast satellites, five different types of audio formats, and seven transponders were used to accommodate each participating station's ability to receive satellite signals.

"That" delay

Radio signals travel at 186,000 miles per second. Communications satellites are positioned 22,380 miles over the equator. A little quick math will tell you that it should take about a quarter of a second for the signal to make the round trip. If the path necessary to make the final "bounce" to earth takes two satellite "hops", there should be about a half second delay. Many radio stations learned this for the first time during this event at Walt Disney World.

Mix Minus IFB

IFB, "Interrupted Feed Back" or "Interrupted Fold Back" (depending on the equipment), is the means by which the on-air talent during a remote broadcast can hear program return of their station's audio. This can be done by many means, from calling the station and being placed on hold, to the program being feed back to the remote location via satellite. By far the most popular method of IFB return is the dial-up telephone circuit that is fed with the program audio which is interruptable by the producer for relaying cues and orders. This configuration allows full two way communication between the remote site and the studio.

Mix Minus IFB is the normal IFB signal "minus" the satellite signal from the remote location. This in effect eliminates the confusing echo caused by the satellite delay of the talent's voice.

CONCLUSION

The 15th Anniversary press party at Walt Disney World was a seven day event which began a year long celebration that includes ongoing radio remotes. The following statistics were accumulated during this press party and includes data from both parks.

Radio

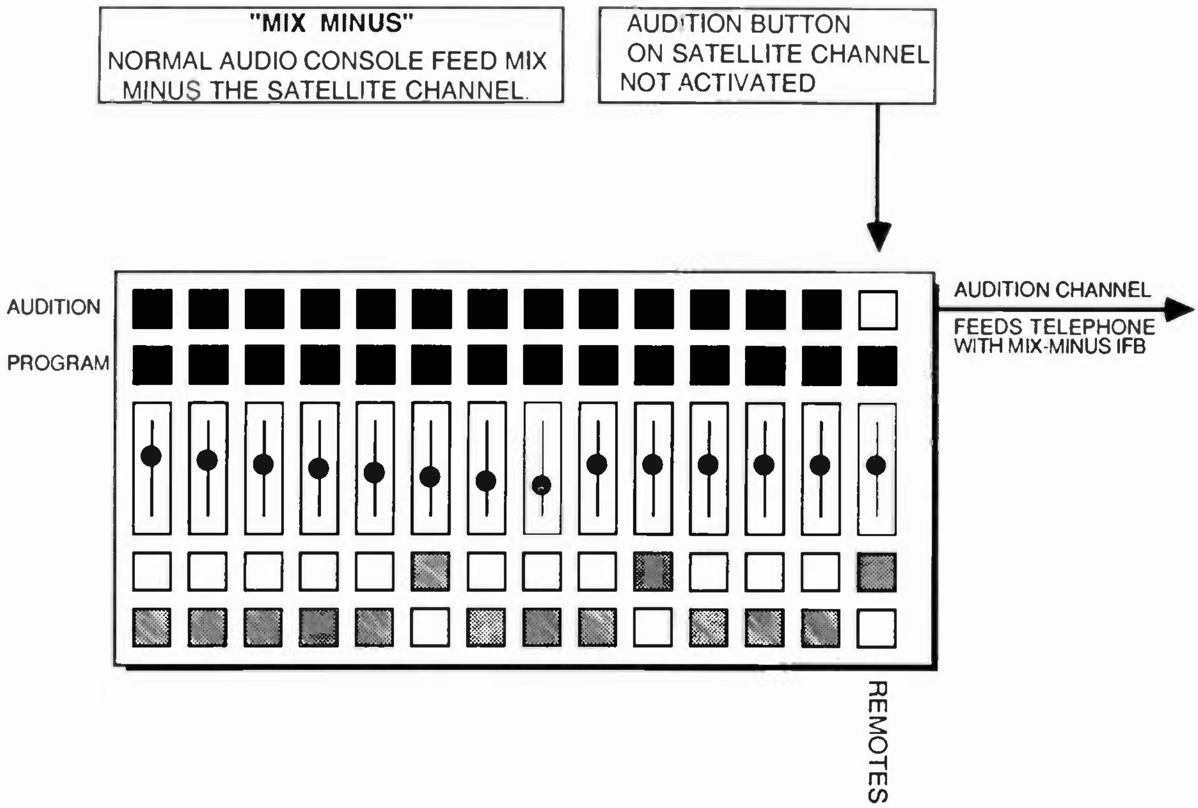
Total Stations	154
Via Satellite	92
Via Telco	62
Max. Simultaneous Broadcasts	97
Total Remote Broadcasts	260
Total Hours of Radio Broadcasts	1005

Television

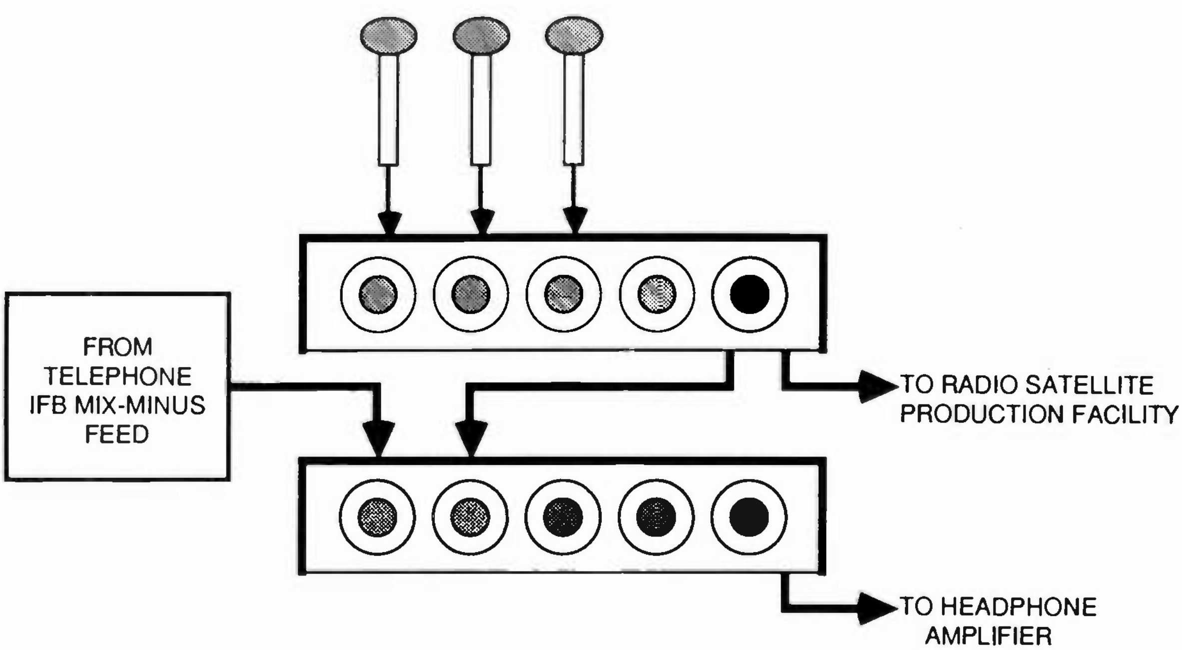
Total Stations	121
Max. Simultaneous Broadcasts	16
Total Satellite Feeds	353

Telecommunications

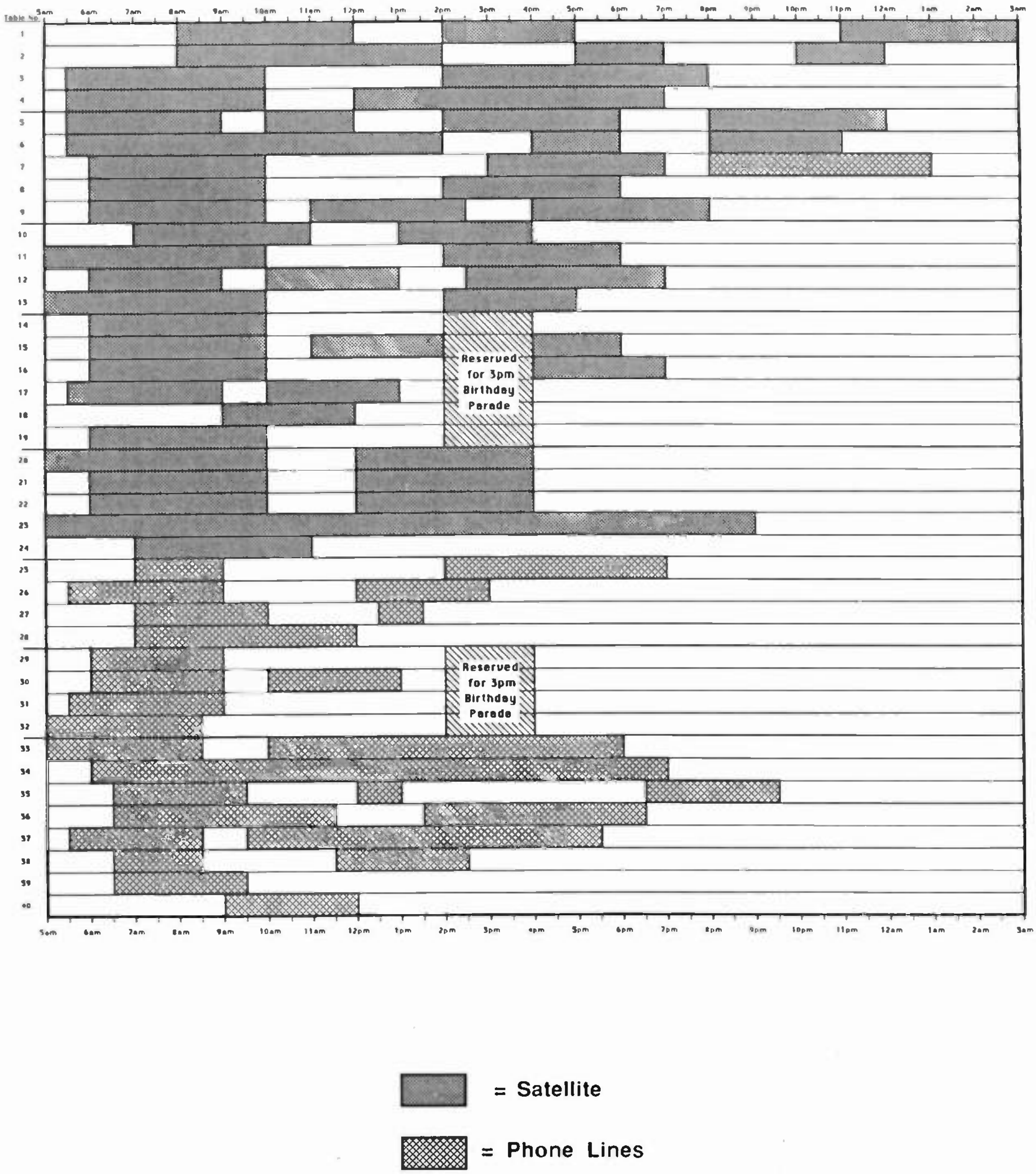
Additional Number of Phone Lines	690
Additional Number of Pagers	480
Additional Number of Cellular Phones	40
Additional Number of Special Circuits	56
Feet of Temporary Fiber Optic Cable	2,500
Feet of Temporary Copper Cable	50,000



Console Set-Up at Studio for Mix Minus IFB



Remote Broadcast Position at Walt Disney World



15th Birthday Radio Schedule for Oct. 3
MAGIC KINGDOM

AUTOMATIC PHASE CORRECTION FOR TAPE CARTRIDGE MACHINES

James R. Carpenter
Broadcast Electronics, Inc.
Quincy, Illinois

Time delay (phasing) errors between the left and right channels in stereo tape cartridge machines causes erratic high frequency loss and other compatibility problems when the channels are "summed" to monaural either before transmission or within a monaural receiver tuned to a stereo broadcast. These problems are created by azimuth errors between the playback head gap and the material recorded on the tape. The common causes of azimuth errors are head misalignment, gap scatter and the time dependent variations of the tape cartridge and its related tape guidance system.

Presently used cartridge machine phase correction systems use a one time alignment during recording to correct for an average phase error value. This average correction only partially addresses the problem since the amount of correction needed changes as the tape cartridge mechanism revolves. Real time correction is required to completely correct phase errors. Stand alone systems are available to correct for phase errors, but they are expensive and require encoding of a reference signal on each tape.

This paper describes a technology that corrects phase errors in real time during tape motion without the encoding of a reference signal.

LIMITS OF PERFORMANCE

In order to set the design limits of the correction circuits, the performance limits of the tape cartridge medium need to be defined. The parameters that we are most interested in are azimuth error and its effects, signal-to-noise, frequency response, separation and distortion.

Azimuth

Azimuth refers to the head gap orientation with respect to the direction of tape travel. Absolute azimuth denotes that the head gap is perfectly perpendicular to the direction of tape travel as shown in Figure 1.

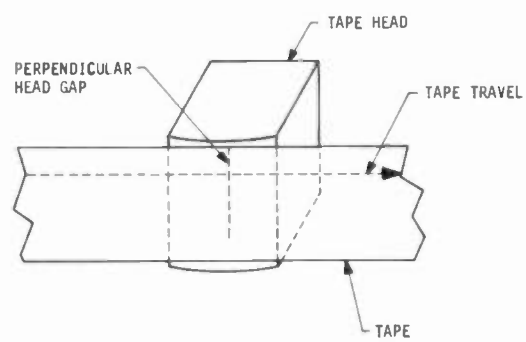


FIGURE 1. ABSOLUTE AZIMUTH

Relative azimuth error depends on the difference between the azimuth of the material recorded on the tape and the azimuth of the playback head as shown in Figure 2. Relative azimuth errors cause two problems in tape cartridge machines. The first is a non-recoverable loss of high frequency information during playback. Figure 3 shows the amount of high frequency loss versus relative azimuth error according to the formula:

L=20 log sin (180 T)/3.14159*T

L= loss in dB
T= tan (A)*F*W/V
A= relative azimuth error (degrees)
F= the frequency of interest (Hz)
W= track width (inches)
V= tape speed (inches/second)

Azimuth errors also cause time delay (phase shift) errors between the two audio channels. This interchannel time delay can cause image shift in stereo systems and cancellation of some signal components in summed monophonic systems. The cancellation effect is most pronounced at high frequencies because the differential time delay results in a phase error which increases with frequency according to the formula:

$Pe=360*Td*Fs$

Fs is the frequency of interest.
Td is the interchannel time delay.
Pe is the interchannel phase error.

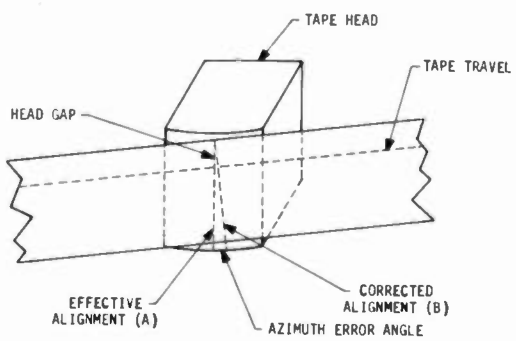


FIGURE 2. RELATIVE AZIMUTH ERROR

From Figure 3 it can be seen that an azimuth error of 0.5 degrees causes a 10.5 dB loss at 15 kHz. This is an interchannel phase error of 40 degrees at 1 kHz, which is equivalent to an interchannel time delay of 115 microseconds.

This 115 microseconds of delay causes complete cancellation of 4.3 khz and 13 kHz components in a mono sum signal. The maximum tolerable system delay error would seem to be less than 16 microseconds. This is equivalent to 90 degree interchannel phase error at 15kHz (relative azimuth error of 0.07 degrees) which gives a maximum stereo frequency response loss of 0.15 dB at 15 kHz. The maximum mono sum loss would then be 3 dB at 15 kHz.

It is obvious from these figures that the onset of monophonic signal degradation occurs with much smaller relative azimuth errors than those which affect the stereo channels only. This information on the amount of error necessary to degrade the monophonic versus the stereo program material sets the necessary time delay correction range. Along with the typical cartridge machine performance data discussed next, it sets the boundaries for the performance of the phase correction system.

Signal-To-Noise

The signal-to-noise of a tape machine is determined by the tape type and operating level. Without some form of noise reduction, an unweighted S/N of 60 dB below 250 nWb/m pulling tape is achievable at this time.

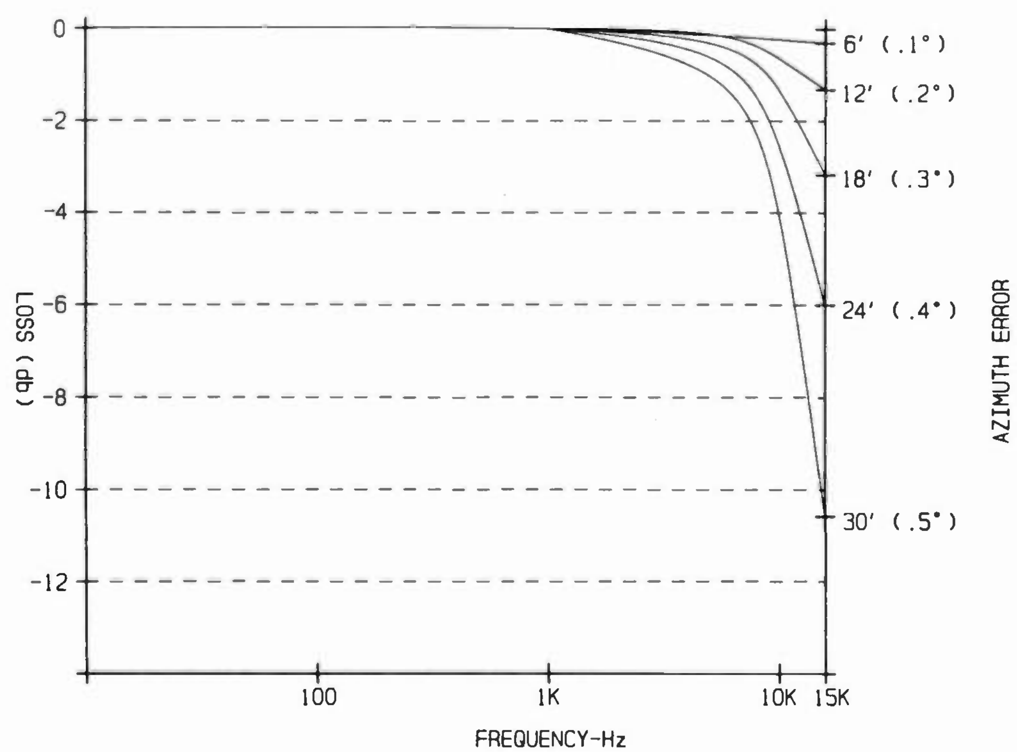


FIGURE 3. LOSS AS A FUNCTION OF FREQUENCY AND AZIMUTH

Frequency Response

The frequency response of a tape cartridge machine is limited by the amplitude of the low frequency contour effects of the head. Newer head designs provide a frequency response of ±1 dB from 30 Hz to 16.5 kHz.

Distortion

The distortion performance of a tape cartridge machine is limited by the tape type and the recorded level. Newer tape formulations allow distortion figures of under 1% at record levels of under 250 nWb/m.

Separation

The separation performance of any tape machine is dominated by the tape heads. While separation performance of 60 dB is possible at frequencies less than 1 kHz, inductive coupling between the coils of each channel limit the separation performance to the neighborhood of 35 dB at 16 kHz.

In order for the phase correction circuitry to be transparent to the cart machine performance, it needs to exceed the performance specifications given below in Table 1.

PHASE CORRECTION PERFORMANCE MINIMUMS

Distortion	<0.5%
Signal-To-Noise	>70 dB
Frequency Response	<±0.25 dB 40 Hz-16 kHz
Separation	>60 dB 30 Hz-16 kHz
Correction Range	>±115 microseconds (> ±620 degrees at 15 kHz)
Correction Error	<±16 microseconds (< ±90 degrees at 15 kHz) (< ±30 degrees at 5 kHz)

METHODS OF PHASE CORRECTION

There are several available methods to correct phase in cartridge systems. They tend to fall into two broad catagories: encoding systems and non-encoding systems.

Encoding Systems

Encoding phase correction systems usually inject some type of control signal onto the tape when it is recorded to define the correct phase relationship of the right and left channels. The advantage of the encoding systems is that the correction circuitry can be optimized for a known reference signal. For example, one system uses a modulated 19 kHz pilot signal recorded on both audio channels. Another system records left channel audio on the cue track for a reference.

While not usually thought of in these terms, a system which uses sum and difference matrixing techniques is also an encoded system. This process uses one channel for sum (L+R) information and the other channel for difference (L-R) information. Each playback machine must have a decoder (dematrix) and each recorder must have an encoder (matrix).

When they are actually used, encoded systems usually give the best phase correction performance. The disadvantage of encoded systems is the need to encode each cartridge in the system to take advantage of the correction capability. In order to actually use the correction performance of an encoding system, the entire cartridge library, all agency spots and any other cart not recorded with the encoding system must be re-recorded. Encoding systems also require each playback cartridge machine to have a decoder assigned to it, which imposes large cost, maintainance and complexity penalties on the entire audio system.

A subset of the encoding phase correction system is a cart machine system that mechanically adjusts the relative azimuth of the record head to that of the playback head during a setup procedure. This allows the machine to correct for the average phase error of that <u>one</u> tape machine/tape cartridge system. This system ignores the questions of machine to machine interchangability, cartridges that were not recorded on that system (eg. agency spots) and changes in tape cartridge phase performance due to wear and minor damage. It cannot correct for real time changes in phase error due to the rotation of the cartridge mechanism, which creates time dependent variations in the amount of tape skew.

Matrix encoding systems do not solve any phase problem, they just introduce the problem in another form. With phase problems in the discrete audio channels, there will be phase problems in the matrixed audio. In matrix form the result is usually poor separation and a poor stereo image for the stereo listener. Since the channel signal-to-noise ratio for a tape machine is fixed, it is also likely that the final discrete signal-to-noise ratio will degrade. Most stereo signals have much more L+R than L-R information. In the worst case (L or R only), the L+R channel will be 6 dB lower than a discrete channel. This is because when L=R the amplitude is twice the single channel value (6 dB). When the decoded noise contributation of the L-R channel is added during dematrix, there can be as much as a 9 dB degradation of signal-to-noise.

Non-encoding Systems

Non-encoding phase correction systems are based on the fact that stereo program material has a considerable amount of monophonic content and that this monophonic content can be used to guide the correction process. As was shown in the section on azimuth effects, degradation of the monophonic content of the stereo program occurs well before any degredation of the actual stereo information.

In order to use the monophonic content of a stereo program to correct for time delay (phase) errors, it is necessary to find a way to extract the time delay information from the audio signals on the tape. Signal theory points out a way by using a valuable property of signals called the autocorrelation function.

The autocorrelation function is a time function of the signal. It indicates the degree to which the signal is related to values of itself in time. This function is obtained by multiplying the current signal by a delayed sample of itself and averaging over the sample time. At zero delay, the signal is multiplied by itself and the value is the signal power. As the amount of system delay is increased the value of the autocorrelation will decrease. The autocorrelation function will always have its largest amplitude at zero delay, except for sine waves which have additional, equal value peaks at multiples of the period.

If a signal is multiplied by a delayed sample of another signal, the result is the cross-correlation function, which represents the amount of common information in the signals. Using the two channels of a stereo system, the cross-correlation will then represent the autocorrelation of the monophonic components. This is the information needed to extract the time delay information from the stereo signals. Using a servo system to maximize the value of the correlation function by linearly delaying the lagging channel, will allow real time correction for interchannel time delays.

Unfortunately, finding the peak of the correlation function requires the evaluation of the function at several points and a search to find the largest peak. This process does not lend itself to cost effective real-time implementation in a servo system. However, knowledge of the performance limitations of the system to be corrected allows the design of real-time signal processing techniques to simplify the evaluation of the correlation function. Because each channel is processed before the correlation, the system design can be limited to detecting the zero point of the function, the central concept of any servo design.

This design allows a low cost, high performance real time phase correction to be designed for almost any stereo system. The design of the signal processing circuitry will be dependent upon the performance characteristics of the stereo source device. Since the optimum signal processing circuitry for each source device type is different, there is no easy way to use this technique to develop a "universal" phase corrector. Each type of stereo source device would require different signal processing circuits. Using one signal processing device would require re-adjustment for each source. If a compromise setting is attempted, degradation of both monophonic and stereo performance would result.

SYSTEM EXPLANATION AND PERFORMANCE

System Explanation

A block diagram of the system is shown in Figure 4.

The right and left audio channels are low pass filtered, then input to linear audio delay lines. The audio delay lines were picked as appropriate technology for a cartridge machine corrector system because they have a linear time delay versus frequency charcteristic. It is very difficult to build analog phase delays that have a linear delay versus time characteristic. The delay line audio performance, while not up to compact disc audio quality, is more than adequate for tape machine use. The harmonic distortion is less than 0.25% and the signal-to-noise is greater than 75 dB below normal signal level. The left delay is fixed at one millisecond, the right delay is variable from 0.8 milliseconds to 1.2 milliseconds. This gives a maximum delay range of ±200 microseconds (±1080 degrees at 15 kHz).

After the delays, the audio is low-pass filtered to remove sampling artifacts and is passed on to the noise reduction circuitry. The audio is also sampled at this point for the phase detector signal processor. After signal processing the signals are correlated, rectified and filtered. This filtered, level shifted signal is used to run the right delay clock, completing the phase tracking servo loop.

System Performance

The Lissajous figure shows before and after correction results for a single test tone of 7.5 kHz in Figure 5. The error is slightly greater than 90 degrees (oval trace) and the correction is better than 3 degrees (45 degree line). Figure 6 shows a before-and-after correction results for a pink noise test tape. These pictures were created by manually mis-aligning the playback head of the machine and then enabling the phase correction.

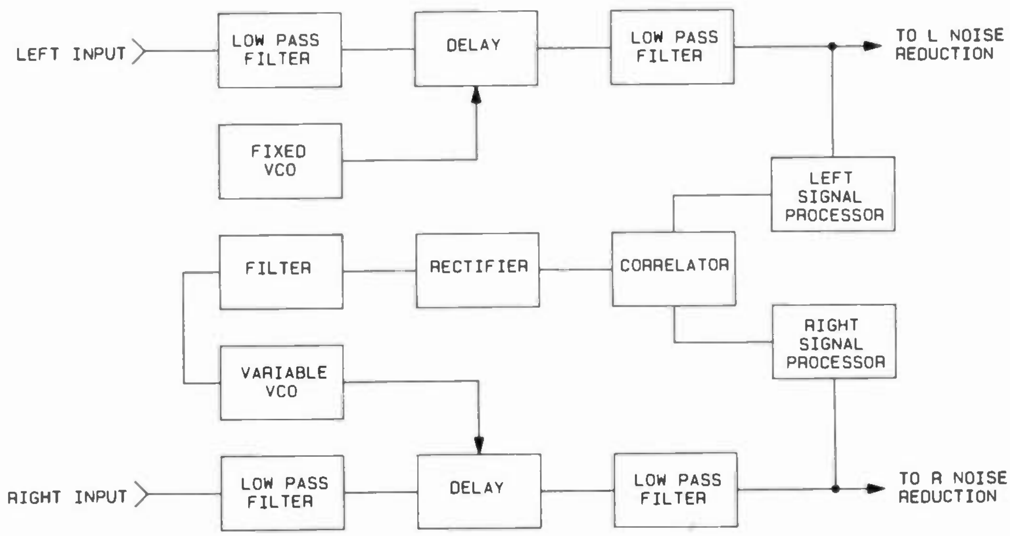


FIGURE 4. TAPE CARTRIDGE PHASE CORRECTION SYSTEM BLOCK DIAGRAM

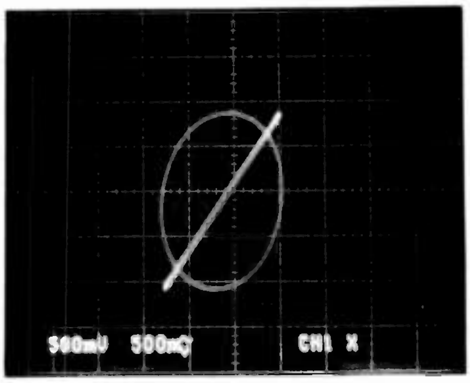


FIGURE 5. TONE PHASE ERROR BEFORE AND AFTER CORRECTION

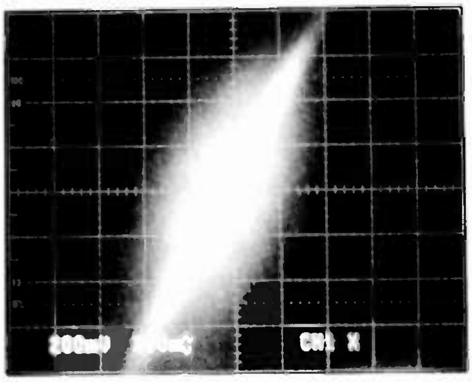


FIGURE 6. PINK NOISE PHASE ERROR BEFORE AND AFTER CORRECTION

The trace in Figure 7 is a five minute sample of relative uncorrected cartridge machine playback phase error (degrees) versus time (minutes). A new 6 minute cartridge was used. It was recorded with a 7.5 kHz tone on the same cartridge machine used for playback. The large excursion inphase at about 4 minutes is the tape splice. The trace in Figure 8 is a five minute sample of the same machine and tape as Figure 7, but with the phase correction in circuit. Without phase correction in circuit, the maximum peak phase error (excluding the splice) is 17 degrees, with short term variances of as much as 15 degrees. With the phase correction in circuit, the maximum peak phase error (excluding splice) is 1 degree, with short term variations of 2 degrees.

This is an obviously an optimum system. It has a brand new cartridge and freshly tweaked record and playback alignment. Figure 9 shows a non-encoded five minute sample of the same machine, but with a different tape that was recorded on a different machine. The phase offset is 80 degrees. The maximum peak phase error is 85 degrees, with short term variations of 10 degrees. With the phase correction in circuit (Figure 10), the phase offset is eliminated. The short term error is reduced to less than 3 degrees.

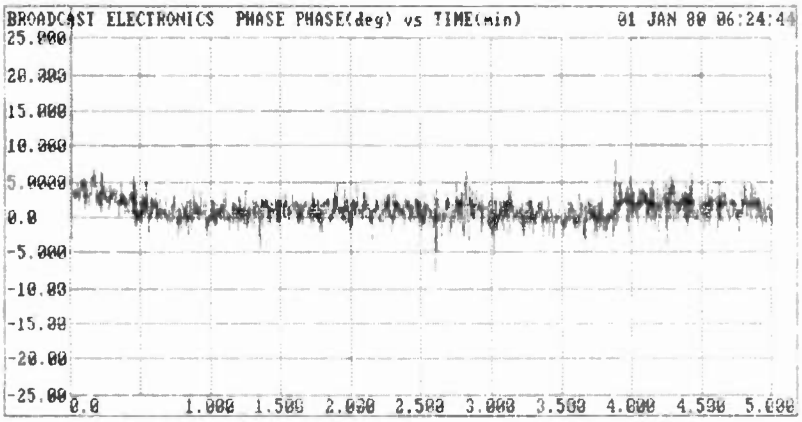


FIGURE 7. UNCORRECTED PHASE ERROR VERSUS TIME

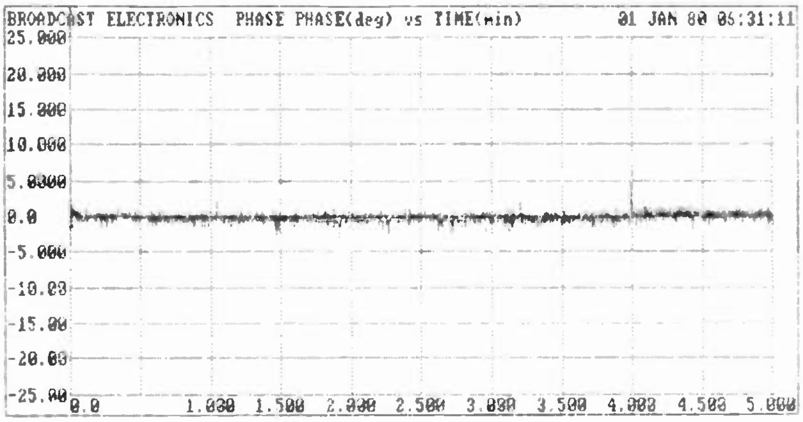


FIGURE 8. CORRECTED PHASE ERROR VERSUS TIME

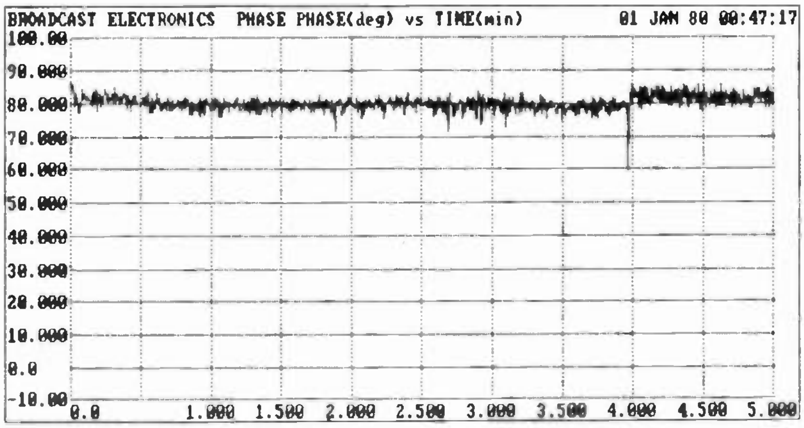


FIGURE 9. OLD CART-UNCORRECTED PHASE ERROR VERSUS TIME

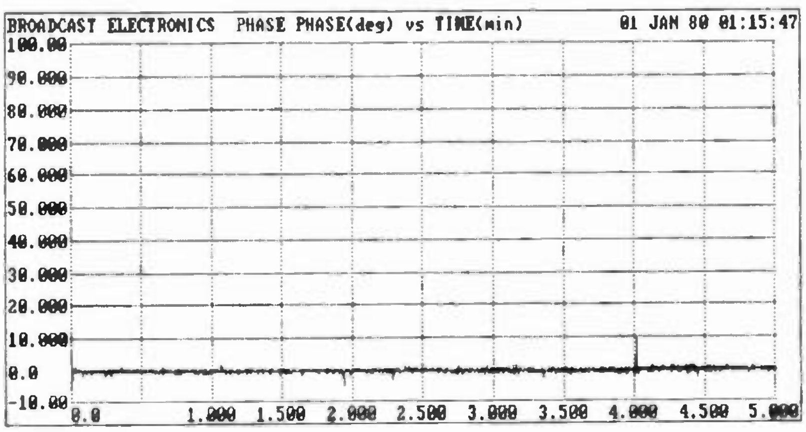


FIGURE 10. OLD CART-CORRECTED PHASE ERROR VERSUS TIME

CONCLUSION

A built-in, non-encoding interchannel phase correction system has been profiled in this paper. This cost effective, operator transparent system eliminates the monophonic compatibility problem for any tape cartridge machines playing any tape, without the inconvience and expense involved with encoded systems.

ACKNOWLEDGEMENTS

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NEW GENERATION AUDIO ROUTING SWITCHER PERFORMS MANY FUNCTIONS

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Crossmatic E(TM) is a microprocessor controlled signal routing system, whose modular configuration and building blocks allow for versatile applications in radio and TV broadcasting studio and network switching. Either electronic crosspoints or relay technique is used, depending on the type of signal to be switched. Primarily intended for audio switching, the system is capable of handling several different signal levels, such as 4 channel audio (stereo, foldback etc.), intercom (2- and 4-wire), data (remote control, time code), and signalization.

Latest state-of-the-art microprocessor techniques control the crosspoints either via conventional CRT terminals or custom built keyboards with pushbuttons and alpha-numeric displays. A large number if subprocessor and interface modules enable the control from peripheral equipment of different complexity as well as from primary host systems (processors or process computers).

Plans for future development towards reduction of space and power consumption software modularity and the final goal of digital audio switching will be given.

1. INTRODUCTION

It is a fact that the number of signals that most radio and TV broadcast stations routinely handle is growing and elaborate switching systems for several signal levels are required. One is aware of the development steps in routing technology, which began with patch panels for signal distribution, still in use today as bypass in emergency situations. Then relays were introduced for remote controlling, which are still not out of fashion for switching signals with e.g. exceptional voltages or frequencies. Reduction of space and cost have been achieved through electronic crosspoints. Nowadays, microprocessor systems offer not only improved system operation but are gradually leading to total automation of the switching process. And finally, one has to envisage digital sound switching. Based on our experience of producing standard and customized routing systems for the last two decades, this paper tries to focus on some present and future aspects of modern routers given by the example of our CROSSMATIC E series.

2. SYSTEM STRUCTURE

One of the main features one expects of a modern routing system is modularity, resulting in high flexibility for adaptation to user requirements and later on for easy expandability. Specifically designed for switching audio, intercom, signalization and data (no video), CROSSMATIC is a fully modular system consisting of a variety of amplifier, crosspoint and control modules. It can purposely be divided into the three main parts for switching, control and operation, outlined in the following items.

2.1 Switching Parts

As far as the switching modules are concerned, one always has to compromise between destination oriented X:1-types and the more economical and more easily expandable x:y-types. We primarily use 16:4 configurations, but several X:1 relay cards are also available. For audio switching the usual chain input amplifier-electronic crosspoint card-output amplifier is used. Blockdiagrams of these basic matrix modules are shown in Figs. 1 to 3.

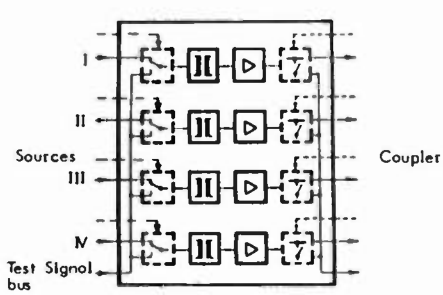

Fig. 1 Quadruple input amplifier PM-4ET

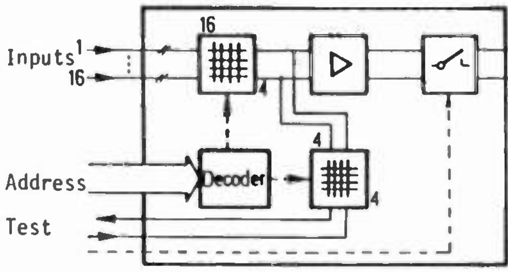

Fig. 2 Electronic audio matrix EAM 16:4/MEM

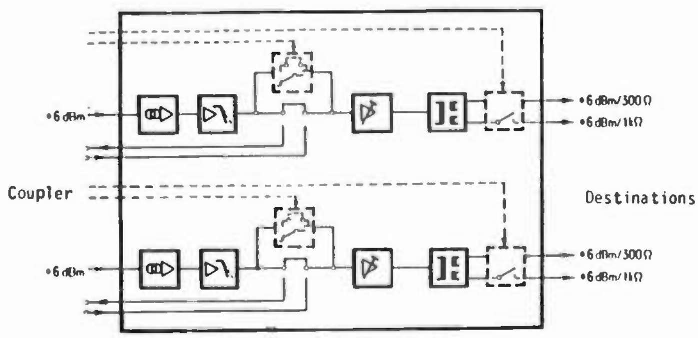

Fig. 3 Dual isolating amplifier PM-2TVE

Ample measuring and monitoring facilities play an important role in station operation. Special attention, therefore, is drawn to some useful features: the inputs amps have an integrated, remote controlled input bus e.g. for feeding test or identification signals into the normal program path, and an output bus for pre-listening the matrix. Similarly, the output amps have input relays for the purpose of shortcircuiting unused outputs to minimize noise contribution, and a separate balanced floating output, thus yielding an integrated mono or fixed stereo monitoring bus without necessitating additional switching modules.

The electronic audio matrix features op-amps with tri-state FET switches at its 4 outputs for unlooped connection of further matrix blocks and to facilitate implementation of 3-stage couplers (see item 3.4). In addition to its four balanced 4x4 CMOS-type crosspoint switches, a fifth one enables a skillful selftest of the card, more closely described in item 3.2 Reedrelay cards having the same 16:4 configuration with 2 or 4 contacts per crosspoint handle virtually all ancillary signals, and an additional version with mixing resistors on card, is especially useful to switch 4 wire intercom with conference character.

2.2 Control System

Today's requirements for the operation of modern routing systems can only be fulfilled by microprocessor techniques, not only in the central control unit (CCU), but also for the data preprocessing at the input/output periphery.

The control hardware of the CROSSMATIC series consists of a comprehensive range of modular units built around the 8 bit-INTEL 8085 family and is shown in Fig. 4. (See top of page 142).

It enables manual control from a terminal or customized keyboard, automatic control at predetermined times from a stored instruction list and external control from a host system acting as master (process computer, processor of a video system, etc). Control of the switching parts is done by means of a special module called Intelligent Rack Interface (IRI) in each matrix rack, which controls crosspoints and amplifier-relays via a parallel bus looped-through to all card frames. For maintenance, an optional set of test modules is available. The modular structured and block oriented software (SW) is written in PL/M or PASCAL, partial ly also in Assembler and then is converted into machine language. The invariable parts of the program, e.g. system monitors, are stored in EPROMS. Matrix status, name tables and error list etc are laid down in non volatile memories e.g. CMOS-RAM which, in case of a power failure, will preserve this data for at least 7 days.

2.3 Peripheral Equipment

On one hand, operators of routing systems are used to the simple operation of yesterdays rather small switching devices, but at the same time expect the high degree of operational comfort offered by todays processing facilities. According to our experience here, we tend as far as possible to recommend the use of data terminals considering all the advantages of a CRT with user dialog and matrix status display. This also seems to be the only feasible choice for the control of middle to big-sized routers. Only for very small switchers, or in cases where only destination oriented parts of bigger ones shall be controlled, or whenever fastest possible operation is the primary goal, push-button panels with LED displays for selecting single sources and destinations (which we will call keyboards) should be used. Quite a lot of different customized keyboards from button per crosspoint, button per in/output up to keypads with "take/enter" button and special function buttons have already been delivered together with suitable peripheral controllers. Their connection to the CCU is mainly done in a starlike manner via 4 wire serial data links (RS232C ports) and with selectable transmission speeds up to 19.2 kbaud. External storage media, e.g. floppy disks and printers for protocol printing complete the list of useful peripheral equipment.

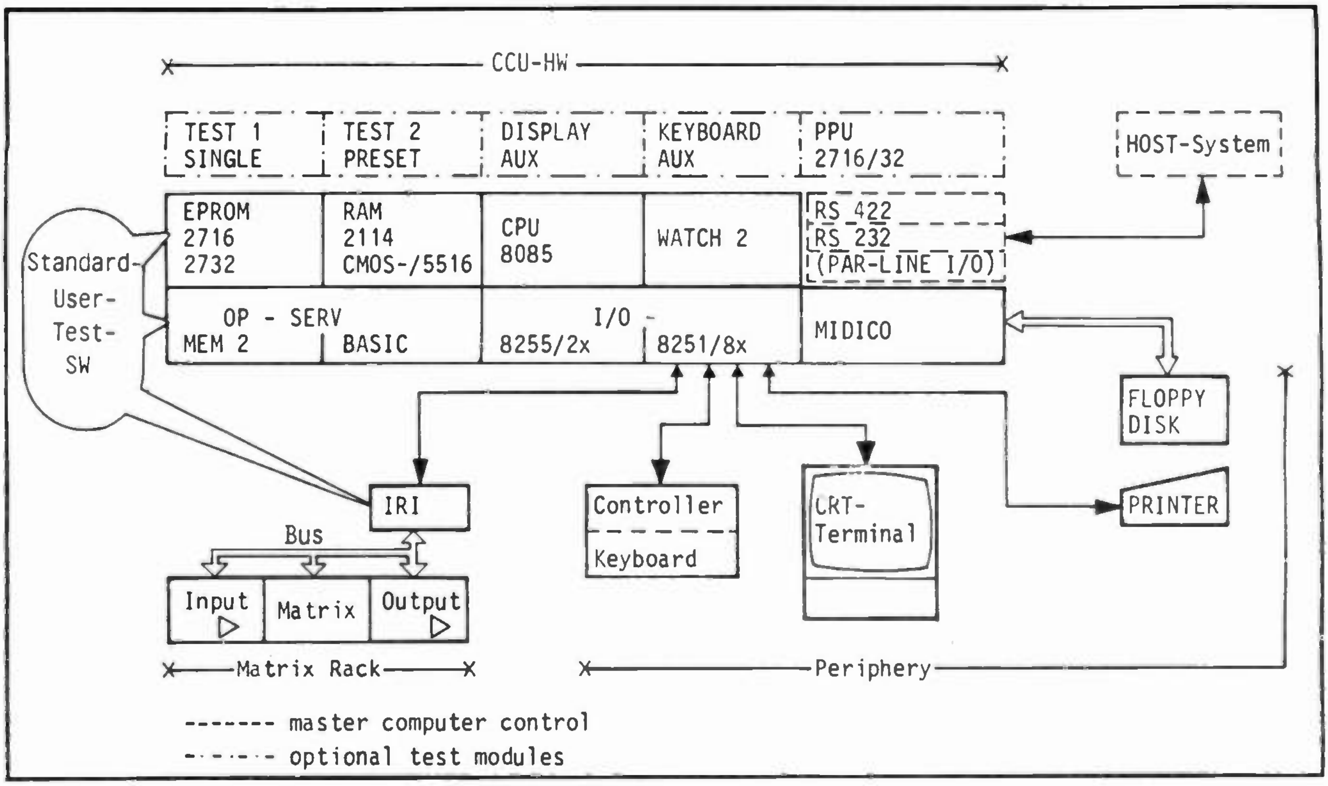

Fig. 4 Control System

2.4 Mechanical Configuration

In spite of the utmost system modularity, costs for engineering and production should be kept low, while maintaining the same high quality standard. Input, output amplifiers and crosspoint modules are separately combined to form building blocks of 32 or 64 input amps, 16 output amps and 768 or 1024 electronic crosspoints in one 3RU card frame each. Rear mounted mother boards for crosscabling instead of individual wiring help to comply with the above requirements for low assembly cost and, for example, constant low crosstalk figures. Interconnection of the individual cardframes is almost exclusively done via ribbon cables (50 core types for audio, 34 core for the IRI control bus and 9 core cables for distributing the various voltage from switched mode power supply units). The card frames are packaged in standard 19 inch racks with or without front doors and hinged frames for mounting cable adapters plus multipin connectors for the external cabling and e.g. special cable distributors as used for 3 stage couplers.

3. SPECIAL SYSTEM FEATURES

3.1 Operation

Operation and control of the routing system can be performed either manually, fully automatically, or from an external host system (master computer). Manual operation is basically covered by item 2.3, indicating some advantages of CRT terminal control and the limited usefulness of individual keyboards for larger routers, in that they become too large, hard to oversee and expensive. While manual operation, even supported by computer control, still requires at least one keystroke, the ultimate goal to ease the burden of the operator is fully automatic operation through a time-controlled processor. This has been realized by means of a command storage preloaded with a certain switching instruction repertoire plus an executing, SW controlled real time clock. Depending on the number of switching instructions per day and the number of days (a one week cycle usually covers 1+7+1=9 days), the data are stored either in CMOS-RAM or externally on a floppy disk. The operator only supervises the procedure with the help of a special display format on the CTR screen. Finally, control from a master computer is realized via a transmission line and an echo line between transmitter (host) and receiver (routing system) following a prescribed transmission protocol. In most cases, a special handshake procedure has been used for assuring error-free transmission and the data have been transmitted via serial RS 422 interface.

3.2 Diagnostics

Some of the most important and therefore expected features of any complex technical system, directly related to its operational availability are notions such as: preventive and diagnostic testing, automatic supervision, error message and localization, preservation of system status, manual and/or automatic by-passing and the like. After switching on, the control system automatically performs certain test functions for proper operation of the memories. During normal system operation a diagnostic background routine, carried out by the IRI module in each rack separately, enables the system to automatically check unused electronic crosspoints.

This "Selftest" is made possible by the use of a separate test IC, which can feed in a test signal for checking the unused crosspoints. In this way, the actual audio paths and e.g. no parallel ones are tested and failure messages are issued in full text, ie. type of failure and rack number etc. In addition, the IRI supervises the power supply units of the matrix and, if necessary, causes a switchover to a stand-by unit. A second RS 232-port on the IRI permits by the connection of e.g. a CRT terminal, the individual checking of a rack during normal operation.

In case of a mains failure matrix status, error memory and name lists etc. are preserved for about 7 days by being stored in buffered CMOS-RAM. Automatic by-pass switching is possible through additional in/outputs in a crossbar-system and even more easily in case of a 3-stage coupler via another free path. Signal paths, which for maintenance reasons are out of function, can be by-passed manually via an optional emergency patch panel. And finally, arriving at maintenance concerns, an optional set of auxiliary and test modules allow for taking the system into operation, for manual input via 24 hexadecimal and command push buttons with separate display unit, for stopping the dynamically running processor at any time or at preset positions of the program checking then step by step in machine cycles and finally, for programming and changing EPROMs by means of a PROM programmer with complete SW support.

3.3 Matrix Levels

In the studio and broadcast business, it increasingly turned out as being useful, to route all type of signals belonging to a certain program production together as one "married bundle". This began with "audio follow video (AFV)" switching in the TV area and nowadays encompasses also multichannel audio (stereo or two channel, feedback for monitoring purposes etc), intercom (2- and 4-wire with and without the ability of building up conferences) signalization (as related to intercom, cue signals, on-air tally etc) down to (digital) data (machine control, time code). As already indicated in item 2.1, the CROSSMATRIC system has all suitable modules and components as well as necessary control facilities available, to handle multilevel matrix systems (with the only exception being video signals) in a married or (partially) separated mode of operation.

3.4 One/two/three-stage Couplers

While routers were getting smaller as modules and block units, they always tended to get bigger as systems and may then be quite expensive and require a lot of rack space. Therefore, early attempts have been made, which aimed at a reduction of the number of crosspoints (as compared to the number x.y of the "one-stage" crossbar switch x:y) while at the same time maintaining 100% availability of the possible connections between given numbers of sources and destinations. The often used "two-stage" coupler consisting of an input and an output distributor with only a certain limited number of cross-connections, in only a compromise, not a really satisfactory solution. Better, and for large-sized routers in certain applications already proven as being useful, is the concept of the "three-stage" non blocking coupler. The principle and one of the realized projects for the German broadcasting station NDR is shown in Fig. 5.

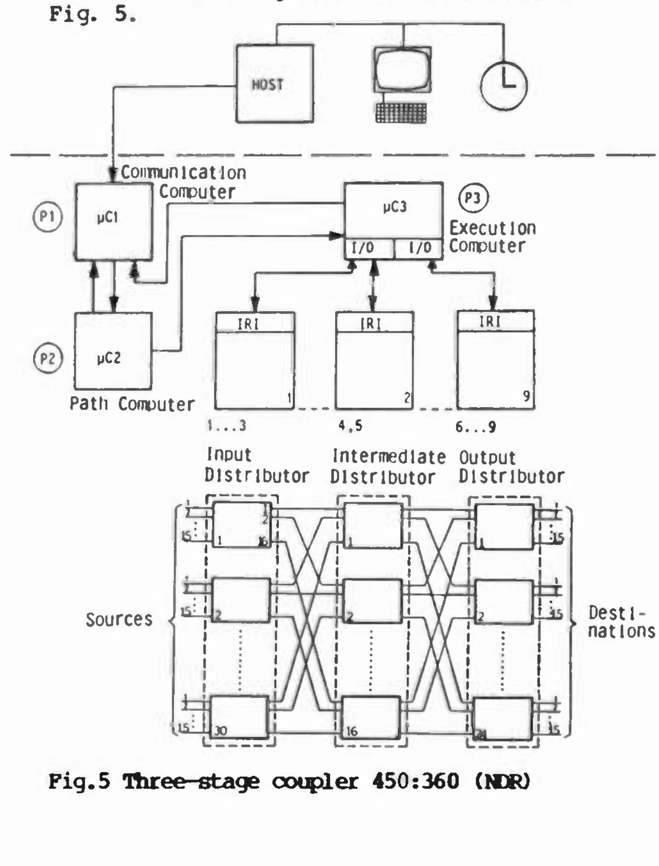

Fig.5 Three-stage coupler 450:360 (NDR)

This concept may lead to a remarkable reduction of crosspoints down to 1/3 ... 1/6 of the number of the comparable "one-stage" (crossbar-) matrix, but of course, also has its drawbacks. First of all, the suitable signal path for each requested connection: source/input-intermediate-output distributor/destination, has to be calculated, thus resulting in a considerably longer reaction time of the system. Secondly, this type of coupler basically is a blocking network, if, as in broadcasting usually and necessary, sources are distributed to more than one destination, so that in a blocking case it has to be totally reorganized. Consequently, a three-stage coupler system is primarily useful for advanced rather than real-time switching operations.

3.5 Specification

The question about reaction and switching times may not be the main point for e.g. production switchers, but can be an important requirement for routers in on-air-switching continuities. Our control concept enables for one-stage matrices and single switching action reaction times (ie. times for executing the whole process beginning with e.g. the keystroke plus data processing plus actual switching) of less than 40 ms, and switching times (ie. the gap in audio signals at destinations, when switching over from one source to another) of less than 10 ms. Furthermore, it is possible to prepare a maximum of 8 so-called "salvos", each one consisting of up to 60 (connection/disconnection-) commands and being executed in typically less than 100 ms.

Concerning technical data, the routing system meets and partially exceeds a certain European standard represented by the so-called 3/2 specs of the Institute for Radio Engineering (IRT) of the Federal Republic of Germany. They are not all given here in complete detail, but two interesting data are as follows:

- peak noise voltage as per CCIR 468-2, unweighted $\leq$ - 88 dBq

- crosstalk attenuation in the range of 40 to 15000 Hz $\geq$ 100 dB

4. VIEW TO FUTURE DEVELOPMENT

Technical systems have continuously to be developed further, so as to adapt to the present state of the art. Based on the success of the CROSSMATIC E series we, of course, do not want to change the basic structure of the system, but primarily to take the main points of the following items into consideration.

4.1 Redesign of Analog Modules

At first we are going to increase the compactness of the various modules for switching analog signals by converting to suitable components for the new Surface Mounted Devices (SMD) technology thus increasing the number of crosspoints per volume of rack space. But, of course, none of the goodies of our present system, such as ample test signal and monitoring busses, diagnostic features and switching rate etc., shall be lost, while at the same time neither loosing emergency patching nor specification figures, but maintaining or even increasing system reliability. We thus hope to arrive at a component density of about 220 electronic crosspoints per rack-unit (RU) with input/output amplifiers, and of about 730 crosspoints per RU without amplifiers, a figure which is quite important for large sized routers.

4.2 Control Concept

After thoroughly studying the market, we are playing with the idea of shifting from INTEL devices to MOTOROLA's 68000 family with programs written in the high level language C. In this way, with 16/32 bit processors and together with VME bus technology, we hope that we can cope with all demands of a large routing system as outlined in Fig. 6 (see top of page 145), especially concerning its control complexity, while at least also maintaining its present switching speed for highly sophisticated configurations.

4.3 Digital Sound

First experiences in switching digital sound have already been demonstrated in 1981, when at the European AES convention in Hamburg, we switched 12 bit nonlinear digitized sound via electronic crosspoints. Thus we proved the insensitivity of digital sound to disturbances and the advantage of unbalanced digital sound switching, that means e.g. stereo signals be switched via a normal analog "two-wire" crosspoint card. Most promising, not only for non-broadcast quality sound such as intercom, seems to be the principle of the so-called "virtual coupler" which, in contrast to the usual multipath crossbar matrices, is a coupler achieving its multiple paths through time division multiplex. This coupling principle is shown by the blockdiagram of Fig. 7.

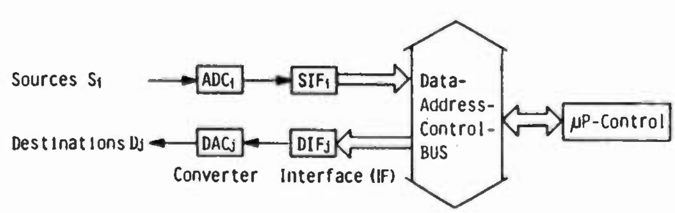

Fig. 7 Principle of Virtual Coupler

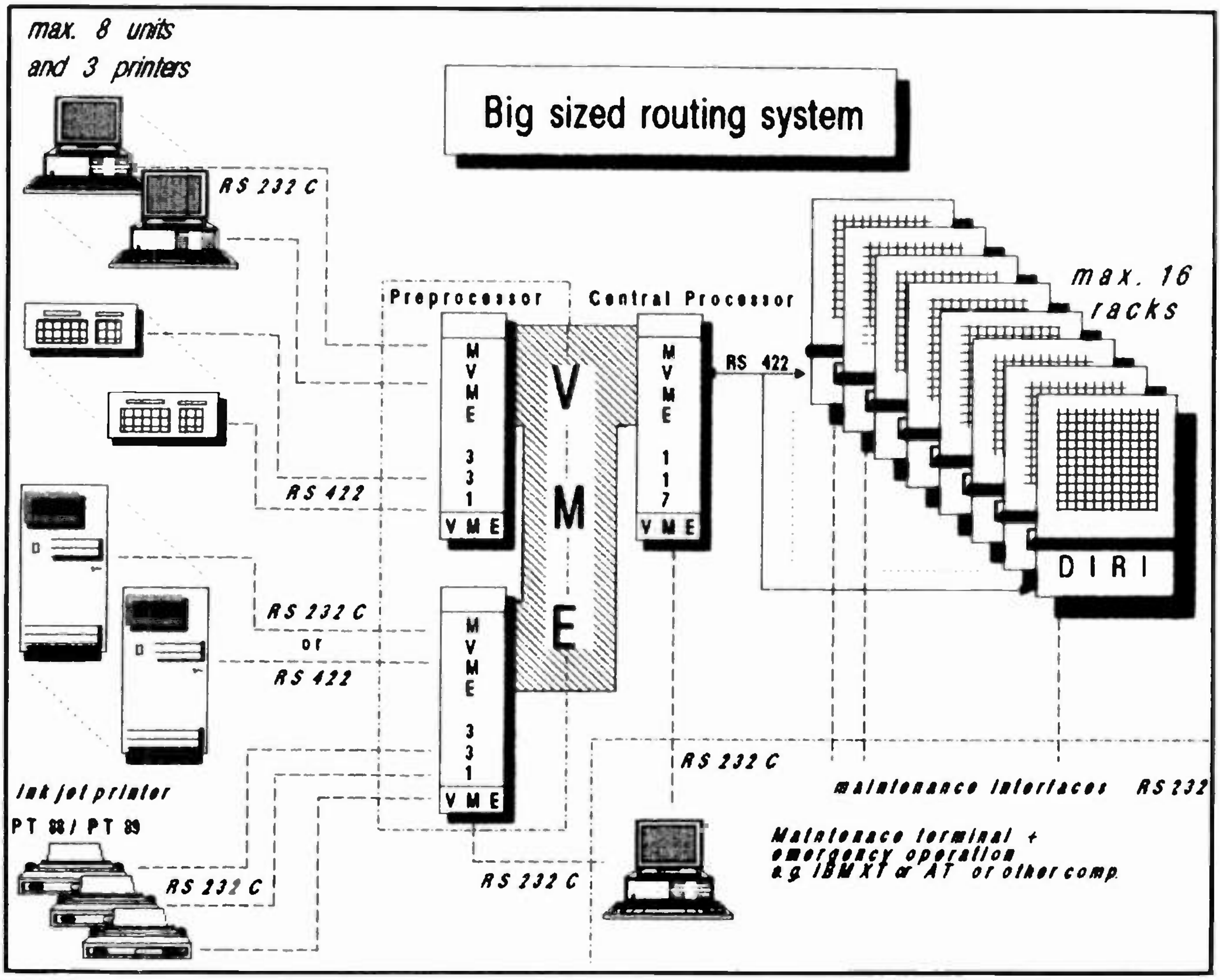

Fig. 6 Big Sized Routing Systems

Here, instead of analog galvanic crosspoints we use the concept of "time-windows" in a time multiplex bus, where the computer sorts out the correct output-data words from the multitude of input data-words on the bus. A PCM coupler for 128 in/outputs with a nonlinear 8-bit resolution and 16 kHz sampling frequency has already been realized, but only yields an audio-bandwidth of less than 7 kHz, thus offering intercom, but not audio quality. High quality digital audio switching still implies a couple of intricacies, some of which are roughly indicated in the next item.

4.4 Future Aspects

Out of the bundle of open questions concerning the development of digital routers, only 2 are dealt with here. As already indicated, some of the most problematic points are some technical data to be expected with digital audio. At present, only 16 bit resolution A/D and D/A converters are economically available. According to the relation

$$S/N\ (dB) = 6.02\ n + 1.76 \rightarrow 98\ dB$$

due to the quantization noise and considering a maxiumum signal level of +22 dBm, the theoretical result is an absolute noise level of -76 dBm rms giving an unweighted peak level of -70 dBq. The difference between the value of -88 dBq given in item 3.5 and -70 dBq theoretically therefore is 18 dB. This will be even worse in practice by another 6 to 8 dB thus resulting in total to some 25 dB of missing S/N ratio. A similar unfavorable situation exists for the nonlinear distortion, where the present requirement for analog audio of a THD value of $\leq$ 0.1 % down to 40 dB below the normal level of e.g. + 6 dBm, can not be met with present 16 bit A/D converters by at least a factor of 10. The result is that for a stand-alone router switching digital sound, at least 19 bit converters are mandatory to meet present analog data. The difference in sampling rates (32/44.1/48 kHz etc.) to be expected for different sources in comparison to any chosen central system rate, due to a missing standard for digital audio, poses another problem that will need to be addressed. One cannot expect here that sampling rate conversion might be able to solve the problem in an economically acceptable way.

SUMMARY

To meet the stringent requirements of todays (broadcast-) station operations, a modern routing system - besides switching as its main task - should offer a variety of useful or even necessary features. It should not only switch reliably, but also fast and there should be no question about diagnostic features being an essential part of system availability as a criterion for the operational usefulness of such a product. The capability to control several signal levels, not only multi-channel audio but also intercom plus signalization plus data, is another indispensable feature. And last but not least, in manual or in fully automatic operation, the system should still offer that which one already expects today:
a maximized operating efficiency thus reducing stress situations as far as possible.
CROSSMATIC E claims to meet all these objectives. The future results of system development will show how far routing systems can adequately cope with the digital challenge.

USING NEW TECHNOLOGY IN RADIO NEWS GATHERING AND PRODUCTION

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ABSTRACT

KNUZ/KQUE radio, Houston, has recently made two major technological improvements in order to increase the efficiency and productivity of its news department. The first was the construction of the first electronic computerized radio newsroom in the city; the second was the installation of repeater transmitters in the news cars to relay studio-quality audio from virtually any location in the field. This paper intends to be a general look at how this technology has helped our operation, and it's potential for application in other facilities.

INTRODUCTION

Although I intend to discuss the equipment involved in our conversion to an electronic newsroom, I would like to emphasize the word "using" in the title of this paper as opposed to "technology", because it is the people in our operation and what they have been able to accomplish using that hardware that I feel is most important. Hopefully, our experiences will be helpful to those of you who may be considering upgrading your own news operation.

HISTORY

Houston is the 5th largest market in the country; in the metropolitan area, there are some 45 radio stations and a population of slightly over three and a half million. As you might guess, the market is extremely competitive. Our two stations play music that could be broadly categorized as "nostalgia". KNUZ-AM broadcasts with a kilowatt on 1230 khz. and features rock & roll oldies from the 50's and 60's, while our class-C FM, KQUE, goes further back to the big band era and also includes some current MOR. Consequently both stations tend to attract a slightly older demographic, mostly 35+. About two years ago, we looked into some research that showed our audience seemed to be more interested in what was going on in the world around them than the average CHR or top 40 listener. Management decided to move both stations, but primarily KNUZ-AM, towards more informational programming. Some years ago, K-NUZ, as it's name implies, had been quite heavily news-oriented. However in the recent period prior to beginning this project, we had not been as agressive in reporting local news. Management's first step, therefore, was to bring in a radio news professional who would be able to objectively oversee the development of a credible organization. Mr. Greg Jarrett joined us as news director in the summer of 1985, and we were fortunate that he chose to actively seek our opinions and suggestions. The engineering department was made to feel like part of the team, and it was this spirit that fired our initial enthusiasm. We decided that we were going to build the best radio news facility in Houston, something we could really be proud of.

GOALS

Our goals were in three areas. First, to improve our ability to gather the news; second, to present it to our audience as professionally as possible; and third, to be able to manage and recall that information in order to get maximum use from it. Our stations have a history of technical innovation; we wanted to continue this tradition by modernizing our newsroom to the state of the art. In the previous few years we had made systematic improvements to almost every area of our facility: for example in 1976 we were the first station to purchase the Harris MW-1, the first completely tubeless, all-solid-state one-kilowatt AM transmitter. When ABC began satellite distribution of their network, we were the first station in town to install a dish. We went through our control and production studios, phasing out older tube-type equipment and converting our two main production rooms to multi-track. But somehow in all of this upgrading over the years, the news department had been overlooked.

WHY COMPUTERIZE?

The first step was to do some research in order to see what was available and at what cost. We felt that some sort of computer system would help the news operation but we weren't really sure what we needed. If you are considering this problem yourself, one of the obvious things you might be asking is "why should I invest in a news computer?" Indeed, we asked the same question. I think our reasons for doing so are comparable to the needs of many other stations. In our case, we wanted a central information source that could be accessed easily. We wanted stories written by anyone in the news department to be immediately available to any other staff member. We felt the ability to easily rewrite stories for inclusion in a later broadcast was important. We didn't want to "lose" stories once they had been written, but archive them for future reference. And here in the 1980's, it is a fact of life that there is just no more flexible way to manage information than with a computer.

WORKSTATIONS

Our existing news area was a large noisy room with clattering typewriters and wire machines. Once or twice an hour, everything had to come to a screeching halt while the news anchor went on the air. It was not the most productive setup to say the least. Our plan was for a modular "workstation" concept where the room would be divided into four cubicles, one in each corner. A central traffic area would allow access and also house the noisy printers. One cubicle would be used as the on-air newsbooth; each workstation would contain a computer terminal and production gear so that any newscast, story or feature could be prepared in it. Each workstation was to be identical, so that a reporter in a rush could dash into any available booth and not have to be concerned with individual quirks of different equipment. Once we decided on this modular concept, everything else started falling into place.

We next turned our attention to lining up the equipment to be used. Each booth needed a workhorse reel-to-reel recorder; we chose Otari MX-5050-B's, which have become sort of an industry-standard production machine. We looked at a lot of consoles and since space was at a premium, we decided on a compact board: the Broadcast Electronics 4M-50. This came stock with only 8 inputs on 4 mixers, but we designed and built a remote input selector that expanded it to up to 20 sources. This gave us enough for all our network feeds, two-ways, etc., and still allowed room for growth. Also planned for each workstation was a cart record/playback deck, timer, cassette recorder, police scanner, aircheck monitor, and the capability for two-way communication with news cars. While the total cost of the project was not our first priority, we did want to hold down expenses where feasible. To this end, we decided not to make the on-air booth a full workstation but to leave in the existing equipment. This saved quite a bit in costs, including the cost of a fourth computer terminal. Once we had our equipment list together, we bought it all at the same time from one source. This enabled us to get a package price, which resulted in additional savings.

THE NEWS COMPUTER

The next step was to select a computer system. Our initial idea was for a simple word processor; but we discarded this approach when we realized that while a word processor might help to write stories, there would be no easy way to index and recall them, which was more important. We then looked at systems specifically for newsrooms and discovered most of them were designed more for TV stations than for radio, and priced out of our reach. We finally found a system called Electronic News Processing, or ENP, from Jefferson Data Systems of Charlotte, North Carolina. There were several important factors that led to this decision. First, ENP used the same BASYS software that we had seen in more expensive systems. Our Associated Press wire would dump directly into computer memory, eliminating the noisy teletype machine and giving us the ability to store, catalog and edit wire copy as well as locally written stories electronically. Second, we were offered a package deal on the software/hardware combination that was considerably less expensive than we could have obtained by buying our equipment from the local computer store. Third, extensive on-site training was included in the purchase price. This is a very important item to consider when selecting a system, as at first few people in your news department will have much computer experience. When some people think of an electronic newsroom, they may think of a "paperless" facility where stories are composed, edited and finally read on-screen. We decided to have the last link to the newsperson be a paper script and not a teleprompter, for a number of reasons. An obvious one is cost, but primarily, the news staff felt they would be more comfortable reading scripts from paper. At least initially, we wanted the system to augment their work, not completely change everything they were used to. We have the capability of adding a display screen in the air booth for this purpose in the future.

We prepared for the installation by first making a detailed diagram of the proposed floor plan, plotting where the audio, power and computer cable runs would go. We worked out an interconnect system where any workstation could share information with the other. This gave the added advantage of being able to put any workstation on the air in the event of a breaking story or equipment failure, as well as a centralized distribution point for networks, air monitors, and other feeds. After rigging up a temporary newsroom and ripping out all the old equipment and wiring, a local carpenter put up partitions, installed cabinetry, air conditioning ducts, and a hung ceiling. We then carpeted the walls and installed the new audio gear; everything was nearly completed when the computer arrived around the first of August. I would like to thank Mr. Bill Ballard of JDS, who trained our personnel and made us confident in our ability to operate the system. Bill has been available to answer questions by phone ever since, which has been a big help on several occasions. When selecting your own system, be sure and inquire about the support provided; it is very important to choose a vendor who will assist you with questions or problems several months or more down the line.

Our system consists of three main units. The central computer is an IBM-AT with 640K of memory, one 30 megabyte fixed or "hard" disk and one high-density floppy disk drive. The hard drive stores all of the information on the system, while the floppy drive is for initially loading the software, and making backups of the database. It communicates asynchronously at 9600 baud with the other two workstations, which are stock 256K IBM-PC's. Any information in the system can be accessed simultaneously from any of the workstations. The system printer is an Okidata Microline-84, which is shared by all three workstations. We also have a modem to connect our AP NewsPower 1200 service, as well as expansion ports should we desire to add other wire services. If future need arises, we can easily add more terminals as well.

POTENTIAL PROBLEMS

Although much has been said about the advantages of a computerized newsroom, there are several disadvantages to consider as well. Fortunately, most of them are manageable. The most obvious disadvantage is the high initial cost; not every station has $20,000 or so to spend on a system of this type. However as this technology becomes increasingly accepted, the price continues to decline. That figure is probably half of what a comparable system would have cost several years ago. In addition, there are less expensive IBM-compatable alternative computers manufactured by Leading Edge, Tandy, and others. It may be possible to work a trade deal with a computer store in your market. Rather than a network of several computers, you might consider a single stand-alone model, particularly if your news department is small. And there is tremendous variety in the cost and complexity of news software.

Besides news, there may be other uses to justify the cost of the machine in your facility such as spreadsheet analysis, billing & accounting, music rotation or form letter processing. At KNUZ/KQUE, an assortment of information in the computer is made available to other members of the staff outside of the news department. For example, AP sends down advance Billboard charts that are given to the program director. Disk jockeys pull feature material about recording artists and daily historical events to prepare their shows. Sales keeps client information in the database. Engineering uses it to write memos and keep address files of equipment vendors. We encourage everyone to learn to use the system, although of course the news staff has priority when needed. The point is that this massive amount of information is kept in a centralized location, in a logical format that allows quick and easy access by a variety of people. For us, this versatility has helped to offset the high initial cost of the system.

Another potential problem may be members of your staff who are reluctant to learn to use the machine. We had a few of these "computerphobiacs" during our conversion, and all it took was some patient training to help them discover that the system was in fact pretty simple, and did not have electrodes implanted in the chair to zap them if they pressed the wrong key. We occasionally run across what I call the "Andy Rooney syndrome", folks who are so used to their old Underwood or Selectric typewriters that they don't want to give them up. Usually though, once they learn a few simple commands and discover how easy it is to edit text with a word processor, they will be amazed at how they managed before. We don't miss the banging and clanking of typewriters in our newsroom one bit.

If the concept of 30 million characters of your most vital information existing only as faintly charged particles of magnetic oxide makes you a little nervous, then you can appreciate the potential for data loss as another problem to be reckoned with. Magnetic media is vulnerable to certain calamities and indeed the four most terrifying words in our technological age may be, "The Computer Is Down". There are various causes for this to happen from time to time, and the best cure is prevention. At KNUZ/KQUE, we have a strictly-enforced policy of "no eating, drinking or smoking in the newsroom". If you've ever had to clean a spilled soft drink out of a console or wipe off a yellow buildup of cigarette smoke, you know how damaging these materials can be. Having them anywhere near a computer is a disaster waiting to happen. Nevertheless, it is good to keep in mind the three golden rules of computing: Number 1, Make backups. 2. Make backups; and 3. Make backups! This is <u>extremely</u> important. Due to the volume of data on our system, making backups is a somewhat time-consuming and tedious chore, but we do it regularly in anticipation of the day when an inexperienced user will somehow manage to reformat our hard disk. Despite the best of intentions, human errors will always occur. Although our system contains a number of safeguards to prevent accidental damage, these are somewhat like high voltage interlocks on a transmitter: they can be defeated if someone really wants to. Another common source of data loss is disturbances in the AC power. Blackouts, brownouts, spikes, noise and lightning hits on the power line can play havoc with your system, and one thing you should include in the budget when purchasing a computer is some sort of power line conditioning. This spans a broad range from a simple $50 surge suppressor, to a $1000 uninterruptible power supply. Today's computer equipment is well-built and reliable; breakdowns are rare, but they do sometimes happen. Due to the complexity of digital circuitry and the sophisticated test equipment required to troubleshoot it, many broadcast engineers are not expert computer technicians. This is best left to factory or dealer service; a dealer may also be able to provide a loaner machine so as not to disable your entire operation while repairs are being made. It would be wise to make contingency plans for handling problems of this nature. We have been fortunate in that we have not had any equipment failures since installing our system. The computer will occasionally lock up after a power outage, but we have been able to cure this every time by the simple act of shutting it off, waiting a minute, and turning it back on again. I recommend that you try this "off/on fix" before calling for service; our experience has been that most minor bugs and glitches can be rectified by this method.

<u>REPEATER TWO-WAYS</u>

In addition to computerizing our newsroom, we investigated other ways to make the operation more productive. Traditionally, two-way radios have been used for years in news cars to communicate with and relay assignments to reporters in the field. If field reporters had tape from an important event, they would usually have to return to the studio, break up the tape, cart the actuality and write a story around it before it could be aired. This sometimes resulted in a delay of up to several hours between the occurance and airing of a story. While we occasionally had on-the-scene reports of breaking stories, the audio quality over the two-way was poor enough to discourage this practice. For "live" coverage of a special event from inside a building, such as a news conference, we were forced to compete for a few existing telephone lines with their notorious lo-fi sound, or order equalized loops. Loops have many drawbacks, including high cost, and the fact that the phone company requires a 30-day notice to install one. We thought there had to be a better way.

In January of 1986, we decided to replace our old news cars as part of an overall plan to give the news staff more visibility. We made sure, for example, that every time a reporter covered an assignment, mike flags were used in case TV cameras were present. This wound up getting us a fair amount of free publicity on the 6 o'clock news. The cars were painted with our logo in bright eye-catching colors, and we wanted to make better use of our two-way radios as well. We considered installing cellular phones, but discarded this idea as being too expensive and not offering the desired improvement in sound quality. After looking at a number of mobile radios, we became interested in the RPT series of remote pickup transmitters from Marti Electronics of Cleburne, Texas. Although mainly designed for fixed position remotes, they had excellent audio specs. What intrigued us most, however, was that Marti also offered a portable hand-held transmitter that could be carried outside the vehicle. High quality audio could be relayed from this hand held unit back to the studio via a repeat link in the news car, allowing the equivalent of television-style Electronic News Gathering techniques to be used for radio. Under normal operation, the mobile unit communicates with the studio base station on F1. The setup for automatic relay operation is shown below in Figure 1; the reporter first parks the car and verifies receipt of a good signal from the base. He then switches both the car's receiver and the handheld transmitter to F2. A subaudible tone decoder in the car's receiver automatically keys the car transmitter on F1 upon picking up the encoded signal from the hand-carried transmitter on F2. The reporter can transmit in a range of up to a mile or so from the car, allowing access to virtually any point. A portable scanner is used to hear talkback from the studio, if desired. Tuning cavities and a coaxial switch perform the necessary antenna diplexing. A bypass switch allows the car radios to be powered without leaving the key in the ignition. Battery drain is negligible when in standby mode and about 4-6 amps when transmitting; the system can operate with intermittent use for several hours without adversely discharging the car battery.

Installing the repeater system has greatly improved our newsgathering ability. Newspeople can now send complete reports, including tape "wraps" from the field with studio-quality audio. These can be done on the air live, or recorded by the anchor and used in a later broadcast while the reporter heads to the next assignment. Although the repeater function is not used often, it has proved invaluable on at least two occasions: reporting election results, and in covering the Challenger disaster from Johnson Space Center. While our stations certainly could have picked up network feeds of these major events, our own on-the-scene reports added a personal dimension to our coverage that would not have been feasible without these technological improvements.

CONCLUSIONS

In the process of modernizing our news department, KNUZ and KQUE faced problems that we think are shared by many others. For us, utilizing new technology in our newsgathering efforts has solved most of these problems. Using a state-of-the-art system has boosted the news department's morale; this coupled with the increased efficiency offered by the computer system has given us an overall increase in productivity. As the cost of computer equipment continues to decline, similar systems will become accessible to smaller stations as well as major market operations.

While part of our expressed interest was to keep our stations on the leading edge of technology, this was secondary to the needs of the people in our department. The technology must improve their performance as it makes their job easier, or else it is just bells and whistles. We feel it has been well worth the effort.

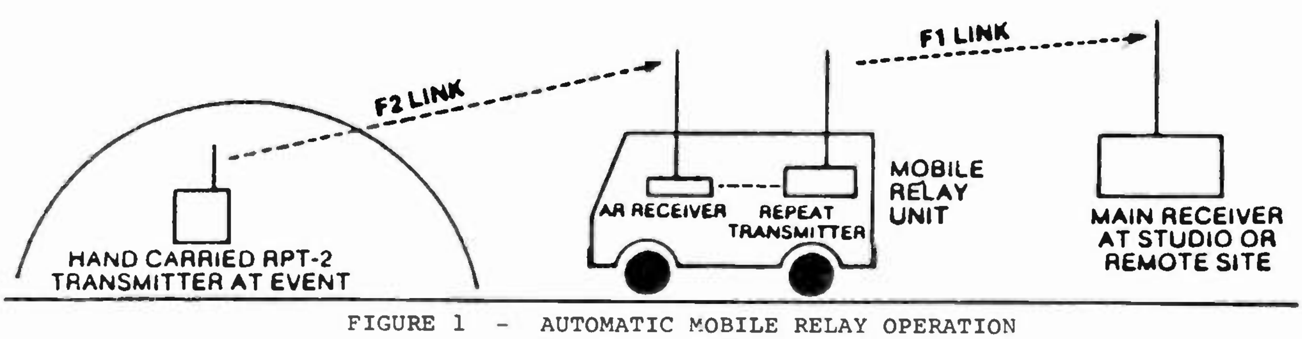

FIGURE 1 - AUTOMATIC MOBILE RELAY OPERATION

A NEW APPROACH TO STUDIO AUTOMATION

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INTRODUCTION

Up until very recent times, audio or video broadcast playback systems usually included a manually controlled mixing and switching console. As the complexity of studio operations rose, these manual control boards also became more complex, thus requiring ever more skilled human operators to work them. Part of this complexity was rooted in the fact that today's studios must routinely handle large numbers of available inputs or program sources, and must also provide a variety of functional options to these sources, as well as to the multiple outputs that are necessary.

The overlay of an additional mechanical controller, or keyboard, to control the new peripherals that utilize an RS232 bus (i.e., RDAT, laser disc, etc.) can in fact create a level of awkwardness that may produce operational errors even when competent, experienced operators are manipulating the equipment. Errors of this nature are, of course, the bane of any studio operation, as they often lead to the unpleasant and unprofitable condition known as "make good," a term no station manager wants to ever be confronted with.

In a radio station, the focus of concentration is in fact on operation rather than "on-air" duties. Therefore, it was desirable to design a control system that could reduce both the hardware and operational complexity, while still accommodating the programming flexibility needed in a modern studio. Touchstone is such a system, and its development has produced a result which meets the most rigorous programming requirements, even though the operator skill level needed is considerably less. In spite of that, the system's advantages are: a greater ease of programming, and much fewer inadvertent mistakes. "Make goods" may in fact become a relic of the past, fading away with the hardware that produced them.

OPERATIONAL OVERVIEW

The core of the Touchstone system is a capacitive touch screen on a highly visible color monitor which acts as the tactile input device for the system. Coupled with that is a unique and proprietary program format, on the screen, that makes interaction with the screen a matter of utmost simplicity. Working with the highly popular IBM PC hardware which has achieved such universal acceptance in the broadcast industry, the Touchstone designers have been able to reduce programming control functions to its simplest human elements. In the computer world this feature has been characterized as "User Friendly;" in the broadcast arena it could perhaps be called "Easy Access Programming."

MACHINE CONTROL

Machine control is accomplished through multi-serial ports at the operator's computer, and all equipment addressable by an RS232 port is started, stopped, etc. For older type equipment we can provide a micro-processor-based mechanical switcher. The Media Touch Systems mechanical switcher provides a basic machine control interface for systems running Touchstone software. The mechanical switcher provides 32 outputs on each unit and up to four mechanical switchers can be daisy-chained for a total of 127 outputs operating from one serial port on a PC.

The mechanical switcher is a micro-processor-based unit with bit-addressable outputs. Each mechanical switcher output is individually programmable and can operate as an on/off or a pulsed output. Pulse width is programmable in 10 milli-second steps from 10/20 milliseconds to almost 11 minutes.

The mechanical switcher provides "closure to ground" remote control outputs for starting and stopping tape machines, and operating relays from a PC that is running Touchstone software. The actual outputs are NPN Darlington open collector outputs and can provide up to 250 mA for driving relays. An optional internal 12 volt power supply can be used to supply power to external relays.

AUDIO CONTROL

All audio mixing and routing is done by a serial interfaced audio mixer, such as Mosley's "Smart Switcher" or Ramko's 16x16 Primus unit. Over 200 audio inputs and outputs can be switched or mixed. We utilize the many outputs for feeds other than the "standard" transmitter feed. For example, level control, cue, mix minus or monitors, telephone, special network feeds, etc. Level or fade control is done from the Touchscreen by corrolation between finger position and audio level, and is programmable by selecting the mid-point decible level. Mid-point can be down 3, 6, 9, 13, or 24 D.B.

We are using a logarithmic digital to analog converter (analog devices 1711 chip). We then ramp the audio level through all intermediate values to achieve the desired decible level.

SYSTEM CONFIGURATION

The complete system has a number of computers linked together to process information and control devices. The hub of the system is a central computer (AT) which is linked to other computers in the system using a LAN (Local Area Network) and provides a central file storage facility.

The network can consist of the following:

1. Editor/Producer's Work Station
2. A Traffic Computer
3. A News Computer
4. A Music Computer

EDITOR/PRODUCER WORKSTATION

The Touchstone Editor/Producer Workstation provides direct communication with the announcer's Touchscreen. This is useful during telephone call-in programs, for example, to send and display messages to the show host about available callers, and to schedule those calls on the Touchscreen workstation.

It is used to perform the operations necessary to generate, edit, print and store (or archive) the daily station operating logs (interacting, if necessary, with an external station billing computer).

Using the Editor/Producer Workstation, changes to the normal operating log are handled with a minimum of disruption. Should the need arise (i.e., an emergency change in spot scheduling due to a tragic news story), authorized station personnel may add, change or delete commercials immediately.

Special broadcast log "templates" (i.e., sporting events, election coverage, severe weather situations, etc.) may be created, then called up and substituted for the normal operating log at any time, without destroying the normal log sequence. Format changes may be created ahead of time and implemented with very short notice.

The Editor/Producer Workstation is used to enter, change or delete live copy (including spots, PSA's and promos) to be displayed on the Touchscreen during the course of the broadcast day. It is also used to assign permanent source/trays of multi-cart, digital audio, multi-CD, etc., to station jingles, sounders and carted features.

The Touchstone Editor/Producer Workstation is used to create or change the system hardware switching functions (including the adding, changing and deleting of the various audio sources), and the setting of the system date and time clock.

And, if that is not enough, you may also use the Editor/Producer Workstation as a stand-alone word processing system, used to generate business letters, written copy for clients when required, maintain mailing lists for other station departments, write station memos, and storm cancellations, among other things.

The nine items in the main menu of the Touchstone Editor/Producer Workstation include:

1. Work with an hourly log
2. Work with high-level switch
3. Work with tray configurations
4. Create extra events for this hour
5. Work with live copy
6. Send message to announcer
7. Read in another day's log
8. Print menu
9. System utilities

TRAFFIC COMPUTER INTERFACE

Touchstone can tie into the traffic computer and receives all scheduled events, then activation by use of the Touchscreen produces an accurate electronic log with verified printed reports in whatever manner or format you require. This pertains to all events aired or not aired, scheduled or unscheduled, added or deleted, with exact times, length and discrepancies. This includes live commercial spots or even length of phone calls for talk shows. Everything is activated by the touch of the screen, therefore, everything is recorded.

ELECTRONIC NEWSROOM INTERFACE

Touchstone has the ability to interface to an electronic newsroom system. With this interface, news text is prepared in the newsroom and down-loaded to Touchstone. The text can appear in the copy window of the central screen or more properly would appear on a separate monochrome, 14", amber screen. This screen uses a large high definition print and the text can be smoothly scrolled by an external button. It operates much the way a TV teleprompter works.

In addition, for stations that do not have a complete newsroom system, a wire capture program is available with Touchstone that allows selected stories, weather, sports, stocks, etc., to be accessed by the copy function.

In the studio, the announcers can touch the copy access field on the Touchscreen and call up whatever text category they wish. At all times, these files have the latest information either because the editor in the newsroom has downloaded from the news system, or the wire capture program has automatically replaced the old with the new. Text categories are user definable and will usually coincide with the station's selected wire service.

Nothing really changes except that the paper system is replaced by an electronic system and the result is dollars saved, and increased efficiency.

There is another important aspect to Touchstone which adds to its utility. All functions we have described for in-studio control, live copy, logging, etc., can also be accomplished away from the studio by taking Touchstone on remote. By adding two (2) 9600 Baud Modems communication can be established to literally run your station with Touchstone technology. Even live copy can be done away from the station.

SUMMARY

The Touchstone System is a new and comprehensive approach to broadcast studio operations which can greatly enhance both the efficiency and accuracy of today's fast-paced, on-air programming. Modern computer technology combined with an intelligent proprietary "format" give the system a variety of distinct advantages over existing studio control systems.

These advantages include:

1. Full integration of all program related operations within a studio.
2. Cost effective and accurate handling of program segments for smooth on-air rendition.
3. Simplification of on air operation for station personnel even while lowering normal error rates.

To the station manager the Touchstone System may represent a radical departure from present studio control systems, however, it brings with it the best potential answer to comprehensive improvements in operations and profitability.

ERROR CORRECTING SYSTEM FOR DIGITAL AUDIO RECORDERS

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A large improvement in quality of recorded audio signals can be attained by converting the analog audio into the digital domain and then recording the signals as digital data. The original analog signal is sampled at a 48 kHz rate with a 16 bit 2's complement converter so that signal-to-noise ratios of over 90 db are realized and distortion is reduced to a level which is unnoticeable. In order to record this digital data on magnetic tape systems, it is necessary to go to higher linear densities and to higher track densities than those required for the equivalent analog signals. Typical densities commonly used in data recording systems are in the area of 10 to 20 thousand flux changes per inch and tape track densities of 16 tracks on quarter inch tape. At these densities, the number of data errors occurring are minimal so that the error correcting code which must be used does not have to achieve a large improvement in error rate. This means that a relatively simple error correcting code can be used and satisfactory results achieved.

In order to achieve reasonable recording times and use a data cartridge for stereo audio, it is necessary to go to 32 tracks on quarter inch tape and to densities of about 50,000 flux changes per inch. At these densities, the recorded bit size on the tape becomes smaller and defects such as tape surface imperfections or signal loss due to dust cause many more data losses and it is necessary to use an error correcting code which has the ability to correct more errors.

Many different error correcting systems are currently being used. A simple system is one in which an even or odd parity bit is added to a byte of data so that if one of the bits in the byte is known to be bad, it can be reconstructed from the other good bits and the parity bit.

Another system is one using Hamming codes. Such systems develop multiple parity bits using combinations of the data bits. Simple Hamming codes can be constructed which will detect two errors and correct one error.

Similarly, Reed Solomon codes have the ability to detect and correct larger numbers of errors but as the number to correct becomes larger and larger, it is much more difficult to implement these codes.

In magnetic tape recording, an additional problem is that a single error burst can be very long, causing a loss of up to several hundred bits. This situation is usually solved by spatially separating the data and parity associated with a given error correcting algorithm. Then the long error burst is corrected by a number of associated correcting systems. Having two error correcting systems back-to-back provides additional correcting capability. This is used in the compact disc audio system and is a cross-interleaved Reed Solomon code. In this case, because of the high volume of systems manufactured, it was feasible to design an LSI chip to accomplish the correction with a low cost of implementation.

A relatively simple error correcting code to implement is one in which simple parity is used to correct a flagged data error and a CRC check word is used to detect and thus flag data errors. One such code is described in U. S. Patent #4,292,684. In this case, the data is put in frames which contain data, parity, CRC, and synchronization. The CRC is used to determine if errors have occurred in its data frame. Data in the frame is associated with data and parity in other frames. If the CRC indicates bad data in a particular frame, this data can be reconstructed from the data and parity in the associated good frames. When bad frames are encountered, it is also likely that clock synchronization will be lost and the recorded sync word will establish resynchronization. This system works well as long as the error rate is low, but with high error rates correction becomes inadequate. This system also requires fifty per cent overhead plus the addition of the CRC word.

Following is a description of a system which accomplishes error correction at very high error rates and requires less overhead than the previously described approach. This new system (developed with the help of 3M experience in the design of digital, multi-track, reel-to-reel recorders) is the key to International Tapetronics Corporation/3M's use of 3M's Data Cartridge in its recently introduced HCDA(tm)

Digital Audio System.

In the previous system it is only possible to correct data using one associated data parity set. This means that if more than one data and/or parity bit in a set is in error the erroneous data can not be corrected. To improve this situation we can place a given bit of data in more than one data set. As long as all the data and parity that is associated with the given bit is different, then the bit can be corrected in as many ways as there are independent data sets. If the independent data sets are small then the overhead becomes large. However, the data sets may be made longer (for example: twelve data bits and one parity bit) so that the overhead will not be exorbitant. This does cause a loss of error correction capability but is more than offset by the improvement in having multiple error correcting sets. Since the data sets are common with each other at only one data bit the sets are orthogonal. As in the previous system the data, parity, CRC and sync are put together in a frame and the CRC is used to determine whether the frame is good or bad. The previous discussion associated bits of data and parity, but when the frame is constructed, the data in a frame can be considered to be the previously described bit. It is then the frames which comprise the associated error correction sets. It is also possible to have subsets of error correction sets in the frames and this can be done to accomplish more uniform error correction hardware algorithms.

In order to get an idea of the error correction capability of the 3M HCDA(tm) Digital Audio System from International Tapetronics, we can determine the probability of error after correction as follows. Let each set consist of "a" data bits with 1 parity bit and the number of orthogonal data-parity sets be "n". For example the set would be b(1),b(2),b(3),......b(a), b(parity) and only one bit b(1), for example, would be common to all the data sets "n". Next we will assume the channel errors are perfectly random and we denote the bit error probability of the channel as "p". That is if $p=10^{-4}$ there is a probability of one error in ten thousand. If we now look at a particular data bit we can say the probability of it being bad is "p". Since this bit can be corrected by its associated data sets we must determine the probability of these sets being bad. Since the probability of a bit being bad is "p" then the probability of it being good is "1-p". Looking at the data sets, the probability of the bits and parity used for correction being good is $(1-p)^a$. This is true since there are a+1 bits in the set and we have eliminated the bad bit. Next the probability of the correcting set not being able to correct because of a bad bit is $1-(1-p)^a$. The probability of "n" sets not being able to correct is then $\{1-(1-p)^a\}^n$. From this one can see that the total probability of not being able to correct is $p\{1-(1-p)^a\}^n$. The expansion of $(1-p)^a$ is:

$$(1-p)^a=1-ap+a!p^2/\{2(a-2)!\}+\ldots\ldots\ldots a!p^{(power)}/\{power!(a-power)!\}$$

If we assume $p^2<<p$ then $(1-p)^a=1-ap$ and the total probability of an uncorrectable bit is $p(1-1+ap)^n=a^n*p^{(n+1)}$. From this we can see that we improve our correction very rapidly as the degree of orthogonality goes up and we lose by the size of the data set-1 raised to the degree of orthogonality. For example if our data set comprises twelve data bits with one parity bit and the degree of orthogonality is three the probability of an error goes from "p" to "12^3*p^4". For a raw error rate of 10^{-4} the corrected rate is $1.728*10^{-13}$.

When data is recorded on tape it is put into a frame structure which contains data, parity, CRC, and synchronization. We can then look at error correction on the basis of this frame structure and in the previous discussion bits can be replaced by frames and the frame error rate improvement will be the same as that determined for bits. Since as was previously mentioned, tape errors can be many bits long, the associated data (frame) sets must have the data and parity sufficiently separated so that any single error will only cause one element in the set to be in error or the degree of correctability will be impaired.

The next problem is to develop an error correcting algorithm which will satisfy the orthogonality requirements and provide sufficient spatial separation for the size of the errors anticipated. If data is grouped in blocks then the size of the block should be consistent with the size of errors expected to be encountered. This size is a function of the track widths used and the linear data density. Obviously it does not help much to have data blocks much shorter than the minimum error size and it is detrimental to have blocks appreciably longer than the typical error size. With playback track widths of the order of .003 inches minimum error size will be in the area of a few mils. At a recording density of 50,000 bits per inch the block size would be in the area of a few hundred bits. Also the minimum separation between associated error correcting sets should be greater than the longest anticipated errors.

Choosing sets which contain 12 data bytes and one parity byte and third order orthogonality we will have 25% parity overhead and we will be required to do one error correction for every four data bytes. A way of developing an algorithm for this situation is to move data through three positions in a row and generate parity at each position. Other data in three other rows is used in the parity generation and the data is different in each row. This is illustrated by:

A	B	C
D	E	F
G	H	I
J	K	L

In this table data A will move to B and B to C. At the same time D moves to E and E to F. This continues for rows G and J. As data moves from column A new data enters column A and data leaves column C. Since there can be no common data in the sets the time to move the data across the rows must be unique for each move. This means that there are variable numbers of storage positions between the data in the rows. For example there might be 13 positions between A and B and 14 positions between B and C. Similarly there will be varying numbers of positions between data in the other rows. Each of these differences must be unique and all combinations of adjacent sums must also differ from themselves and the individual differences. The data entering column A must also have spatial separation and if A=Data(0) then D could equal Data(4*m+1), G could equal Data(4*n+2) and J equal Data(4*p+3) where m, n and p are integers. The next data into A would then be Data(4). In this case then, data moves into the first column position, is delayed by 4*m+1 and moves into the first position of row 2, then is delayed by 4*n+2 and moves into the first position of row three. Finally it is delayed by 4*p+3 and moves to row four. Since the 4*m,n or p has 1,2 or 3 added to it and the data shifts four times between parity generations it can be seen that each data byte will appear only in one row and will be used only three times to generate parity. Also no two data bytes will appear together in more than one set if the row spacings or delays are appropriately chosen. There are many possible spacings which will satisfy these requirements and following is one selected for its associated properties. The differences or delays are shown.

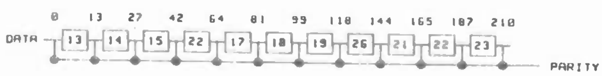
A (16) B (20) C
(13)
D (17) E (21) F
(14)
G (18) H (22) I
(15)
J (19) K (23) L

In order to determine the actual data set from this table it is necessary to remember that the data in a row must increment by 4 and therefore each difference must be multiplied by 4 to find the actual data set numbers. Starting with A=0 the data set would be as follows:

0	64	144
13	81	165
27	99	187
42	118	210

Reorganizing the set in ascending order:
0 13 27 42 64 81 99 118 144 165 187 210

We can show how the data bytes would shift through a delay system to generate parity by:



Starting data positions are shown at the top and the data is shifted to the right with parity bytes generated every fourth data shift. The numbers in the boxes indicate the number of data positions or delays between the data used for parity generation.

The next task is to find an appropriate parity position to be used with the above set. It turns out that if the sets are positioned in the order they will be used for correction, an appropriate parity number is 239. This maintains orthogonality and keeps good spatial separation in the correction sets.

When recording data such as digital audio signals, it is desirable to put the data in frames and construct the frames in such a manner that all the frames are associated with each other in an identical manner. This then allows the error correcting system to handle the error correcting tasks for all frames the same. If the data to be recorded is for computer applications, it will most likely be put into data blocks and the error correcting algorithm developed on a block basis.

Looking at the audio (or continuous signal case) we must determine how many words of data to put in each frame. Since there are three parity words for every twelve data words it is convenient to make the frame some multiple of twelve words. Also since frame error detection is best accomplished with a CRC check word the longer the frame the lower the overhead the CRC adds. From our previous discussion however, we would like our frame length to be in the order of magnitude of the error size. If we choose a recording density of 50,000 fci, then the error size is in the order of a few hundred bits. A good block size would then be 24 data words or 48 data bytes. This then requires 6 parity words or 12 parity bytes in the frame.

The CRC size must next be determined. We need to choose the CRC so that it is consistent in detecting errors to the same degree as the error correcting system is capable of correcting them. The final error rate is a function of the uncorrected errors we expect to see in the recording system we design. If we expect a worst case raw error rate of one in two thousand frames then from our earlier equation, we will achieve a corrected frame error rate of 1.08*10^-10. Since there are 48 data bytes, 12 parity bytes plus CRC in our frame the frame length is a little over 500 bits long. Therefore the number of good bits between uncorrected errors is about 500 times the reciprocal of the corrected error rate or there are about 4.6*10^12 good bits between errors. If the digital audio sampling rate is 48 kilohertz with sixteen bits per sample, then in a stereo system the bit rate is 48,000*16*2 or 1.536 megabits per second. Since we have 48 data bytes or 384 data bits in a frame, the frame rate is 1.536*10^6/384 or 4000 frames per sec. Then we can see that the time between uncorrected errors is the reciprocal of the corrected error rate divided by the number of frames per second. This calculation gives about 2.3*10^6 sec. or 640 hours between errors. In determining the number of the CRC parity bits we will assume that the probability of it determining errors is two raised to the power of this number. The problem with the CRC is not one of detecting errors but one of saying a frame is good when it really is bad. The chance of this occurring is again two raised to the power of the length of the word. We would like to have the capability of CRC detection equivalent to the correcting power of our error correcting system and since this, for our example, is one in 640 hours we should not have an error undetected in that time. Since our raw error rate is one in 2000 or 2 errors per second then 2^x must be about 640*3600*2=4.6*10^6 where x is the CRC length in bits. If we choose a CRC length of 24 bits we find our capability of not missing errors is one in 2^24 or about one in 16 million. This is slightly more than required but is a good choice since it is three bytes long and fits well into our frame structure.

We can now look at an error correction strategy that should be relatively easy to implement. If we look at a byte implementation and if we desire to correct on an absolutely constant frame basis, we must organize the data and parity in such a fashion to accomplish this. Since it is only necessary to perform a correction at one fourth the data rate, and we have 48 bytes in each frame, we need to correct 12 times each frame. Also, since the audio is digitized in 16 bit words, we should use a constant set algorithm for every two data bytes. Following is a table showing the relationship between data and parity to accomplish a uniform correcting algorithm. Data and parity subscripts indicate their position in a frame and the number in brackets is the frame number.

P0[239]=D0[0]+D2[13]+D4[27]+D6[42]+D0[64]+D2[81]+D4[99]+D6[118]+D0[144]+D2[165]+D4[187]+D6[210]

P1[239]=D1[0]+D3[13]+D5[27]+D7[42]+D1[64]+D3[81]+D5[99]+D7[118]+D1[144]+D3[165]+D5[187]+D7[210]

P2[239]=D8[0]+D10[13]+D12[27]+D14[42]+D8[64]+D10[81]+D12[99]+D14[118]+D8[144]+D10[165]+D12[187]+D14[210]

P3[239]=D9[0]+D11[13]+D13[27]+D15[42]+D9[64]+D11[81]+D13[99]+D15[118]+D9[144]+D11[165]+D13[187]+D15[210]

P4[239]=D16[0]+D18[13]+D20[27]+D22[42]+D16[64]+D18[81]+D20[99]+D22[118]+D16[144]+D18[165]+D20[187]+D22[210]

P5[239]=D17[0]+D19[13]+D21[27]+D23[42]+D17[64]+D19[81]+D21[99]+D23[118]+D17[144]+D19[165]+D21[187]+D23[210]

P6[239]=D24[0]+D26[13]+D28[27]+D30[42]+D24[64]+D26[81]+D28[99]+D30[118]+D24[144]+D26[165]+D28[187]+D30[210]

P7[239]=D25[0]+D27[13]+D29[27]+D31[42]+D25[64]+D27[81]+D29[99]+D31[118]+D25[144]+D27[165]+D29[187]+D31[210]

P8[239]=D32[0]+D34[13]+D36[27]+D38[42]+D32[64]+D34[81]+D36[99]+D38[118]+D32[144]+D34[165]+D36[187]+D38[210]

P9[239]=D33[0]+D35[13]+D37[27]+D39[42]+D33[64]+D35[81]+D37[99]+D39[118]+D33[144]+D35[165]+D37[187]+D39[210]

P10[239]=D40[0]+D42[13]+D44[27]+D46[42]+D40[64]+D42[81]+D44[99]+D46[118]+D40[144]+D42[165]+D44[187]+46[210]

P11[239]=D41[0]+D43[13]+D45[27]+D47[42]+D41[64]+D43[81]+D45[99]+D47[118]+D41[144]+D43[165]+D45[187]+D47[210]

Since we have developed an absolutely constant algorithm we can now determine all the other data parity relationships by adding the appropriate numbers to the frame numbers in the previous table. Also we can see that all corrections will be made by performing the correcting algorithm on every fourth data word in each frame. Let's look at an example of how a particular data byte D0[0] will be corrected. The data can be good and not require correction or it can be corrected by one of the following data sets.

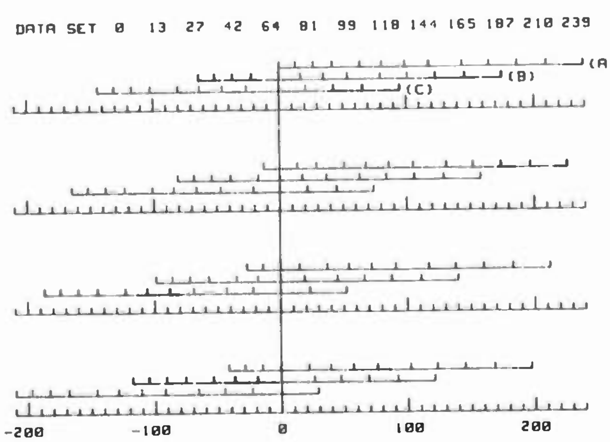
D0[0]=D0[0] (NO ERROR)

D0[0]=P0[239]+D2[13]+D4[27]+D6[42]+D0[64]+D2[81]+D4[99]+D6[118]+D0[144]+D2[165]+D4[187]+D6[210] (A)

D0[0]=D0[-64]+D2[-51]+D4[-37]+D6[-22]+P0[175]+D2[17]+D4[35]+D6[54]+D0[80]+D2[101]+D4[123]+D6[146] (B)

D0[0]=D0[-144]+D2[-131]+D4[-117]+D6[-102]+D0[-80]+D2[-63]+D4[-45]+D6[-26]+P0[95]+D2[21]+D4[43]+D6[66] (C)

Error correcting sets for other bytes may be similarly found by using the foregoing data parity relationships and offsetting frames by the appropriate numbers. Another way of looking at how the error correction is accomplished is to look at the frames spread out and view the associated correcting sets. This is shown in this graph.



DATA PARITY RELATIONSHIPS

Since correction needs to only be done every fourth data word in a frame there are four different correction sets shown. The top set shows correction for the first, fifth and ninth words. The second set for the second, sixth and tenth. The third for the third, seventh and eleventh and the bottom for the fourth, eighth and twelfth words.

The next problem is to determine how to record the data on the magnetic tape. If we try to record NRZ data we find we have a bad low frequency problem if there are a large number of sequential zeroes in our data. The recording system is unable to record D. C. signals so we must choose a bandwidth limited system. Also we must be able to reconstruct our tape clock during playback so that tape speed perturbations during record or playback will not cause signal decoding errors. If we desire a saturation recording system i.e. one in which we only record positive or negative tape saturation, we must constrain the time between successive transitions.

Recording codes that accomplish this are currently called d,k codes where d is the minimum and k the maximum number of cells between transitions. During reproduction the tape clock is regenerated, and data detected by looking for flux transitions on the tape. The highest data rate required for recording is approximately the frequency whose period is twice the time between successive tape transitions. This means that for the same maximum recorded frequency the size of the window allowed for transition detection during reproduce is the time between transitions divided by one plus d. The decrease in window size means that for a given error immunity, where noise is the error cause, it is necessary to increase the system SNR to achieve the same error rate. On the positive side it can be shown that the coding efficiency can be increased by making d larger. However, in a tape system running at maximum density we have found the best compromise is to make d=0. The k was selected on the basis of the lowest frequency requiring good equalization and the ratio of (k+1) to (d+1) is effectively the ratio of the maximum to minimum frequency in the recorded bandwidth. If this ratio gets too high intermodulation and clock recovery during speed perturbations become a problem. For the above reasons a d,k code was chosen with d=0 and k=4. The size of the code was taken to be 8 to 9. This means there are 256 possible input combinations to be coded into 9 bit blocks. With the 0,4 d,k constraint there are still more than 256 9 bit combinations and so some are not used. The ones that were used were chosen on the basis of low frequency content in the 9 bit block. The maximum D.C. content of the blocks turned out to be 22%. Also since in audio the signal will usually be low in level, the code was arranged so that D.C.content was lowest for low amplitude audio words.

In the actual 3M HCDA(tm) Digital Audio System hardware, data is stored in memory to generate parity. The data and parity are then used to generate the CRC word and then sent with the CRC word to the 8-9 converter. A synchronizing signal is then added and the frame recorded on tape. The synchronizing signal is one which violates the constraints of the d,k code i.e. it has a longer time between transitions than allowed by k. It is also made D.C. free by repeating the violation for both directions of magnetization.

On reproduce, the signal is amplitude and phase equalized to return it as close as possible to what it looked like before recording, but with limited bandwidth. A special phase lock loop which can handle tape speed perturbations regenerates tape clock and the signal is decoded back to NRZ. In conjunction with the synchronizing signal the data is converted back to bytes and errors are detected by regenerating CRC and comparing it to the recorded CRC. The data and parity are put into playback memory and error correction is performed on the basis of the CRC comparison.

After data has been stored for a sufficient time for error correction to take place it is ready for output. When this is audio data the output goes to a D. to A. converter to change it back to an analog signal.

COMMENTARY QUALITY AUDIO (7 KHZ) FOR BROADCASTERS ON THE NEW ISDN DIGITAL SERVICE

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ABSTRACT

A new CCITT Recommendation G.722 "7 kHz Audio Coding Within 64 kbit/s" has been developed to provide high quality telephony over end-to-end digital connections. While teleconferencing and similar telecommunication services are the primary application, the new coding provides quality comparable to existing commentary audio services. The improvement in audio quality compared with conventional telephone coding is dramatic. The technical and operational advantages of all-digital channels, along with the widespread introduction of switched 64 kbit/s ISDN service, may make this coding the method of choice for 64 kbit/s audio communication, including commentary quality communications. Internationally, the broadcast application is under study by the CMTT.

THE NEW STANDARD

At the July 1986 meeting, the CCITT Study Group XVIII "Digital Networks Including ISDN" unanimously accepted a new Recommendation G.722 "7 kHz Audio Coding Within 64 kbit/s" for accelerated approval procedures.[1] The recommendation specifies a sub-band adaptive differential pulse code modulation (SB-ADPCM) method of coding which can be generally applied to any telecommunication service carried over 64 kbit/s end-to-end digital switched or private line channels.

As digital channels become ubiquitous with ISDN, a major improvement in telephone transmission quality can be realized, since the new method more than doubles the usable audio bandwidth. While the principle coding rate is 64 kbit/s, coding at 56 kbit/s and at 48 kbit/s is also specified in order to allow the simultaneous transmission of data for such purposes as teleconference management or image transmission. Compatibility with existing PCM coding is achieved by specifying that terminals also incorporate the G.711 PCM standard, which is today's conventional coding method.[2] A companion draft recommendation, G.72Y, specifies procedures for compatible interworking between 7 kHz and conventional terminals, as well as the procedures for establishing the simultaneous data modes.

OBJECTIVE PERFORMANCE

The recommendation specifies several analog performance parameters for the entire circuit, analog input to analog output. These specifications include:

— Attenuation versus frequency according to Figure 1.

— Noise less than -66 dBm0 over 50 to 7000 Hz.

— Single frequency noise at all frequencies less than -70 dBm0.

Other specifications cover the analog sections (including A/D or D/A conversion) of the coder and decoder separately. These include:

— Attenuation versus frequency according to Figure 2.

— Group delay distortion according to Figure 3.

— Noise < -75 dBm0.

— Signal-to-total-distortion ratio according to Figure 4.

The predominant impairment resulting from the sub-band Adaptive Differential Pulse Code Modulation coding is distortion due to quantizing noise, slope-overload noise, and numerical noise in digital processing elements. Figure 5 shows the signal-to-distortion ratio of an encoder-decoder pair, for a single frequency signal. It is characteristic of ADPCM coding that the signal-to-distortion ratio will depend upon the spectrum of the signal being coded, and the ratio will be poorer for wider bandwidth signals.

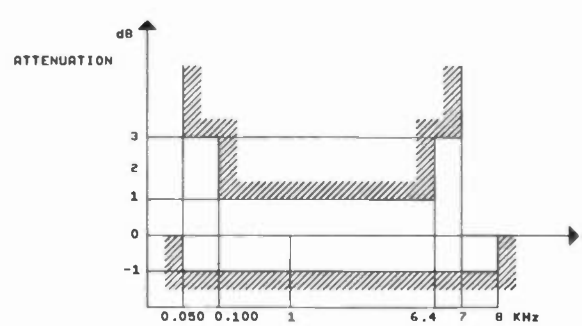

FIGURE 1
END-TO-END ATTENUATION DISTORTION

SUBJECTIVE PERFORMANCE

Since the objective performance of the coders is difficult to relate to the perceptions of listeners, subjective testing must be done to evaluate true performance. The experts of Study Group XVIII, in conjunction with the experts of Study Group XII "Transmission Performance of Telephone Networks and Terminals", conducted subjective tests of voice performance in several countries and languages.[3] The summary results of these Mean Opinion Score tests are shown in Figure 6.

The results demonstrate that the subjects score 64 kHz ADPCM coding very close to 240 kHz linear PCM coding of the same 7 kHz bandwidth test sentences. The scores decrease slightly when the rate is decreased to 56 kbit/s, and considerably more

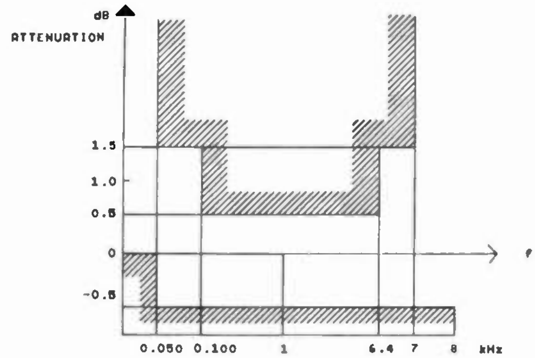

FIGURE 2
PER-CODEC ATTENUATION DISTORTION

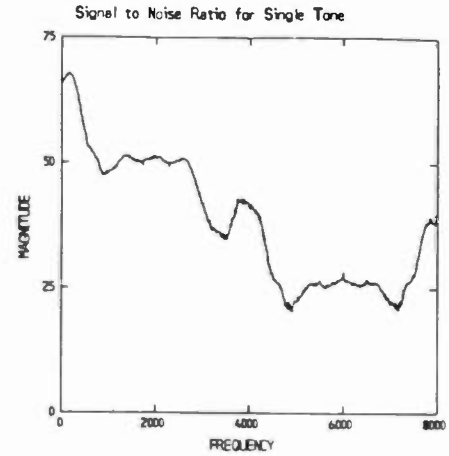

FIGURE 5
END-TO-END SIGNAL-TO-DISTORTION VERSUS FREQUENCY

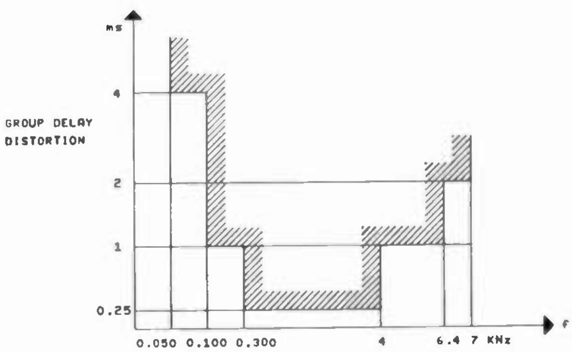

FIGURE 3
PER-CODEC GROUP DELAY DISTORTION

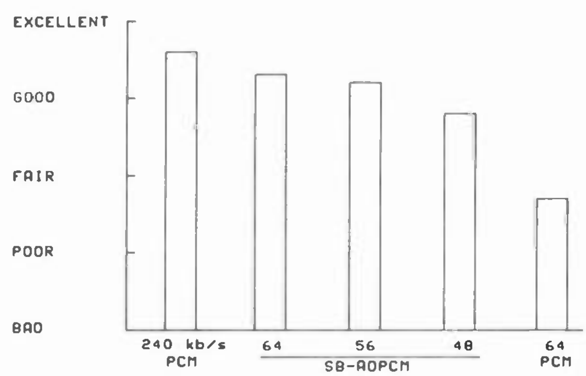

FIGURE 6
VOICE SUBJECTIVE TEST RESULTS

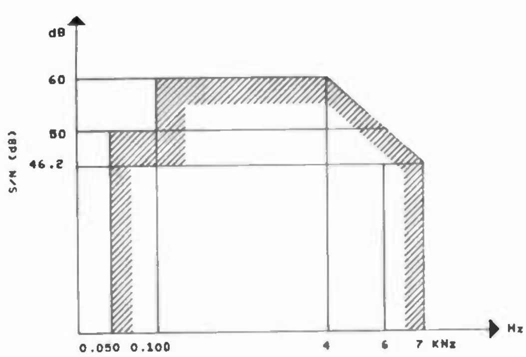

FIGURE 4
PER-CODEC SIGNAL-TO-TOTAL DISTORTION FOR ANALOG PARTS, SINGLE FREQUENCY TEST

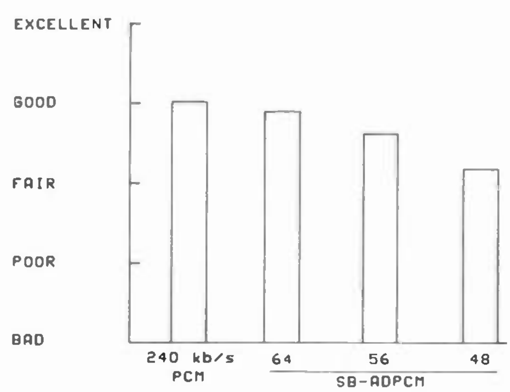

FIGURE 7
MUSIC SUBJECTIVE TEST RESULTS

when 48 kbit/s is used. In all cases, the scores are substantially higher than those for conventional telephone-quality PCM coding. It should be remembered that scores within a subjective testing experiment are relative, and that while G.711 PCM coding scores well compared to analog telephony, or to ADPCM at 32 kbit/s, it scores poorly when the subject has a comparison with 7 kHz coding using the new ADPCM standard in G.722.

A limited test of music was carried out, and the results are shown in Figure 7. The reduction in scores due to the coding is more pronounced. This is probably because the algorithm used was optimized for speech. It should be noted that the music samples were classical, choral, and folk music works.

IMPLEMENTATION

Figure 8 compares today's digital coding method for speech, G.711 PCM, with G.722 7 kHz ADPCM. Using G.711 coding, the input audio signal is band passed to between 250 Hz and 3400 Hz. The frequencies from DC to 250 Hz are attenuated in order to eliminate noise due to power induction, to allow for low frequency rolloff in digital channel bank transformers, and to remove any DC bias in the analog signal. Since 7 kHz ADPCM is intended to be used in an all-digital system, there is no need to eliminate power induction at 60 Hz. Therefore, the low end pass-band edge of the anti-aliasing filter for G.722 is specified to be below 50 Hz.

The analog signal in conventional telephony is sampled at 8 kHz and converted from analog to digital at a quantization level equivalent to 14 bits linear PCM. Each sample is compressed from 14 to 8 bits using a logarithmic function. This has the effect of maintaining the signal-to-noise ratio at a nearly constant value over a wide variation of signal powers. The 8 bits are clocked out at the sampling rate, and the resulting 64 kbit/s digital signal is transmitted over the digital facility.

The audio input to the ADPCM encoder is converted to a 64 kbit/s digital signal in the following manner. The analog input is low passed to 7 kHz, sampled at 16 kHz, and encoded into a linear 15-bit PCM code. Once in the digital domain, the signal is split into a high band (from 4 kHz to 7 kHz) and a low band (from DC to 4 kHz) by a quadrature mirror filter (QMF).

The high band signal is compressed from 15 bits to 2 bits, while the low band signal is compressed from 15 bits to 6 bits. The technique used to compress the two bands, ADPCM, is the same technology that is used for low-bit rate voice coders. ADPCM coding exploits the high correlation in speech samples to actually predict the next sample. It then transmits the difference between the prediction and the actual sample.

If the codec is also transmitting data along with speech, then the data overwrites either one or two bits in the low band. The resulting 2 bits from the high band and 6 bits from the low band are multiplexed together to form an 8 bit word. This 8 bit digital word is output at 8 kHz, resulting in the 64 kbit/s digital rate.

The decoder works in a very similar manner, except in reverse. The input digital stream is demultiplexed into the high and low bands, and data is removed from the 8 bit word if in a simultaneous voice and data mode. The high band and low band are decoded separately, as in the encoder, using ADPCM technology. The two 15 bit digital signals are then multiplexed together into one digital spectrum in the receive quadrature mirror filter. The output of the QMF is converted from a digital to analog signal, and the analog signal is passed through a reconstruction filter. This filter eliminates spurious high frequencies that are the product of the digital to analog conversion.

Bellcore has built its own hardware prototype 7 kHz ADPCM codec. [4] The codec implements both conventional G.711 PCM and 64 kbit/s 7 kHz ADPCM. The codecs use audio equipment instead of a typical telephone handset. The analog input is a

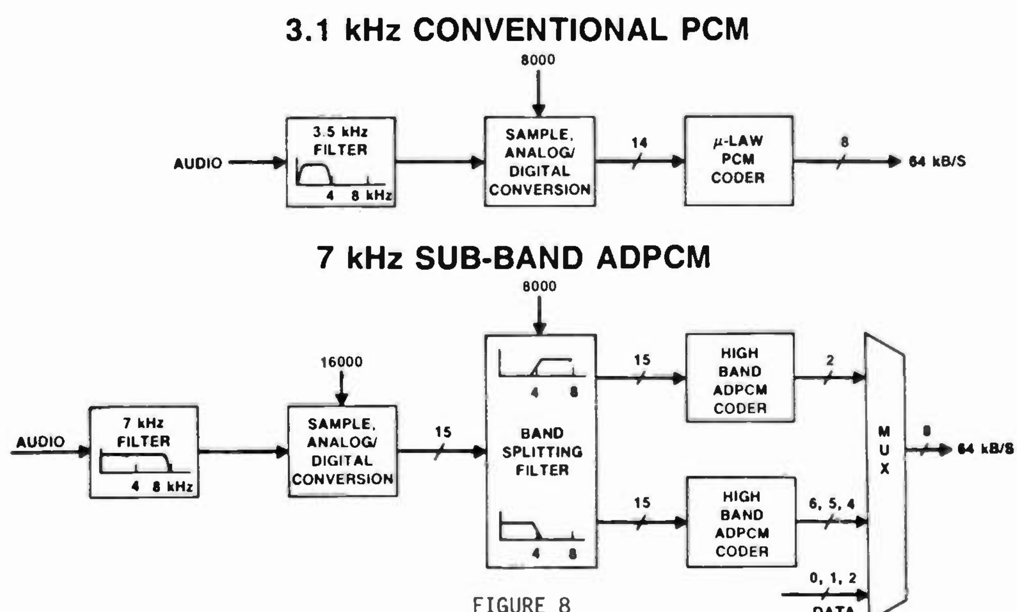

FIGURE 8
CODEC IMPLEMENTATIONS

microphone or a tape deck, and the analog output is through a high fidelity speaker. Switching between modes on the transmit side forces the receiver into the same mode automatically. Most listeners conclude that the improvement over G.711 PCM is quite impressive.

BACKGROUND ON ISDN

The Integrated Services Digital Network constitutes the target architecture for the evolution of today's networks towards ubiquitous, end-to-end digital telecommunications networks, and offers a framework in which both voice and non-voice services can be efficiently and economically provided with new technology. An important fundamental part of the plan is that ISDN extends the 64 kbit/s digital channels inherent in today's digital facilities and switching systems all the way to the customer's premises. At the customers premises both transmission and signaling interfaces are defined so that switched connections can be established between 64 kbit/s interfaces, and so that the connection between the interfaces will be bit transparent.

Figure 9 shows ISDN Basic and Primary Access architectures. The Basic Access provides 2 "B" channels at 64 kbit/s for circuit-oriented transmission, and a "D" channel at 16 kbit/s for signaling messages and packet-oriented transmission.[5] The connection to the network is over a single twisted pair facility. The Primary Access provides 23 "B" channels and a 64 kbit/s "D" channel over 1544 kbit/s facilities. This may be over two pairs using T-Carrier techniques, or multiplexed further for transmission over fiber-optic systems.

The functions provided by the NT1, Network Termination 1 are:

- line transmission termination,
- layer 1 line maintenance functions and performance monitoring,
- timing,
- power transfer,
- layer 1 multiplexing, and
- interface termination.

The functions provided by the NT2 include:

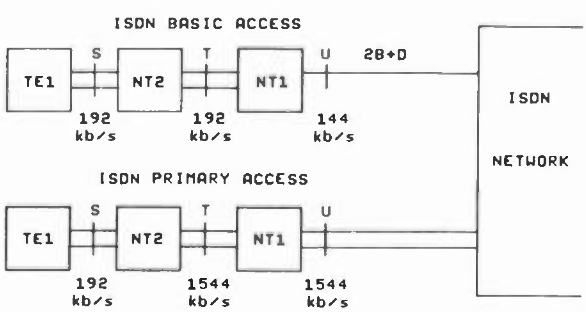

FIGURE 9
ISDN NETWORK ACCESS ARCHITECTURE

- layers 2 and 3 protocol handling,
- layers 2 and 3 multiplexing,
- switching,
- concentration,
- maintenance, and
- interface termination and other layer 1 functions.

NT2 may encompass, for example, the functioning of a digital PABX.

The TE, Terminal Equipment functions include:

- protocol handling,
- maintenance functions,
- interface functions, and
- connection function to other equipment.

It should be noted that in simple terminal applications, the architecturally distinct boxes may be physically combined. The interfaces labeled S and T are well defined internationally, and in the United States the U interface is being standardized as well.

ISDN APPLICATIONS

Figure 10 shows ISDN used for a remote news-gathering pickup. The microphone at the remote is connected to audio equipment, which is then connected to a combined terminal equipment. The terminal equipment contains the G.722 codec, as well as a keypad and signaling equipment to place a call over ISDN. The call is placed to a similar terminal at the studio, using either the 64 kbit/s unrestricted or 64 kbit/s restricted classes of service. The G.722 coding always provides at least one bit having the value of "1" in each byte, and therefore meets the "ones-density" requirements of some existing network facilities which require a minimum ratio of ones to zeros in order that regenerative repeaters and multiplexors can extract timing from the bit stream.

If the remote location had a pre-existing ISDN service, the remote crew could plug their G.722 terminal into the standard S or U interface jack, dial the studio, and begin transmission. No special network circuits would be required, and no special coordination with the network organization would be needed. The dialed connection can be used for two-way communication. High quality transmission will be easy to achieve since:

- the digital ISDN loops will be more immune to noise and power induction than conventional program audio circuit loops,
- the digital ISDN loops do not require conventional analog audio equalization,
- the terminal and the audio equipment can be DC coupled for flat low frequency response, and
- the interface between the terminal and the audio equipment can be engineered to conventional audio impedances and voltage levels.

Thus, the end-to-end performance of the 7 kHz 64 kbit/s connections may compare favorably with the end-to-end performance of today's commentary quality program circuits.

If the remote location did not have an ISDN service, then one would need to be ordered and installed. As ISDN grows, this type of service order would involve common and ordinary operations by the network organization. Both the broadcaster and network organization will have a simpler job since:

— the ISDN access will be installed and maintained by arrangements common to all ISDN lines,

— delays in installation due to the unavailability of "radio pairs" or multiple channels on the digital carrier systems can be minimized, and

— all audio levels, equalization and pre-emphasis are under control of the broadcaster.

These advantages, along with the fact that switched as well as private line services will be available, will allow a quicker response to broadcaster needs, and reduce special training and administrative burdens on the network organizations.

PRE-ISDN PRIVATE LINE APPLICATIONS

Compatibility has been a main theme in Bellcore's participation with the CCITT on G.722. Special care has been taken to insure that the coding algorithm would be usable both in today's pre-ISDN telephone network, as well as in ISDN.

For example, since the North American T-carrier systems do not allow an all-zero 8 bit word, the 7 kHz algorithm has no all-zero code. Secondly, it is proposes that the protocol governing the mode switching capability of such a codec specify that the system always starts up in conventional PCM, and only switch over to the 7 kHz ADPCM modes after a hand shaking operation that insures another 7 kHz ADPCM codec does indeed exist in the receiving terminal. If the transmitting codec does not make this determination, then they both continue to use conventional telephone PCM for the remainder of the call.

After looking at the question of compatibility with current digital private line services, such as DDS, we have determined that it appears that the North American network can support 7 kHz ADPCM at the full 64 kbit/s mode using DDS secondary channel equipment. Although there will be occasional loop-back and other maintenance related false signals in the bit stream, the disruptions they cause will be very brief (much less than 10 msec. in almost all cases) and occur about once a week. The listener will hear a faint pop or a click, and nothing more. This is because the maintenance codes built into the DDS secondary channel codes are very robust in nature, and require that the maintenance codes be repeated every other digital word in order for the system to respond. Thus, the probability that these specific codes will be repeated in this way for any length of time is extremely small.

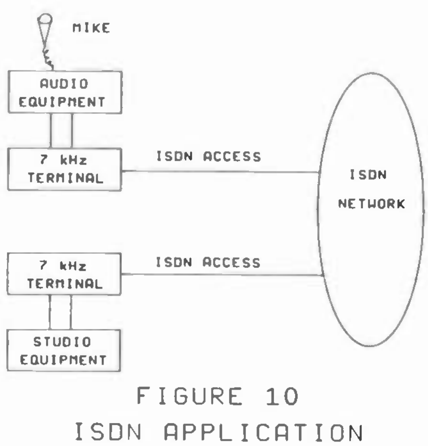

FIGURE 10
ISDN APPLICATION

Said more simply, even before ISDN becomes available, users could use a 7 kHz ADPCM codec with current point-to-point digital private lines as long as the terminal interface is compatible with today's DDS with secondary channel.[6] This will provide the user with the full benefits of all-digital 7 kHz ADPCM listed above.

CONCLUSION

The use of the CCITT recommendation G.722 ADPCM coding in ISDN will greatly improve access to quality audio transmission. Because of end-to-end digital connectivity, many problems associated with analog transmission, including power induction and cable pair equalization, will be eliminated. As the all-digital system spreads, many of the logistic headaches, both for the user and the telecommunications network, are reduced, since the quality audio service becomes just another digital connection.

In fact, the new coding method can be used to raise the general level of telecommunications transmission quality to levels compatible with the needs of broadcasters. We expect that the 7 kHz ADPCM coding method discussed here will not only be used in special private services, but because of its clearly superior audio quality compared to that of conventional PCM coding, it will become commonplace in ordinary business and home digital telephones as well.

REFERENCES

[1] CCITT Study Group XVIII - Report R26, COM XVIII - R26, August 1986.

[2] CCITT Recommendation G.711 "Pulse Code Modulation (PCM) of Voice Frequencies" Volume III, Fascicle III.3, Malaga - Torremolinos, 8-19 Oct 1984.

[3] CCITT Study Group XVIII - Report R17, COM XVIII - R17, April 1986.

[4] "Use of CCITT G.722 7 kHz 64 kb/s ADPCM for Digital Program Audio", T1Y1.1/86-038, Bell Communications Research, July 30, 1986.

[5] CCITT Recommendation I.411 "ISDN User Network Interfaces - Reference Configurations" Volume III, Fascicle III.5, Malaga-Torremolinos, 8-19 Oct 1984.

[6] Bell Communications Research Technical Reference TR-NPL-000157, "Secondary Channel in the Digital Data System: Channel Interface Requirements", Issue 1 September 1985.

SOLVING TYPE C VTR INTERCHANGE PROBLEMS

Larry S. Jefferson
The Public Broadcasting Service
Alexandria, Virginia

The Public Broadcasting Service (PBS) is the largest television network in the world, providing quality programming and related services to 317 non-commercial stations serving the United States, Puerto Rico, the Virgin Islands, Guam and Samoa. PBS does no in-house production, but instead receives programming from member stations and distributes the program material to all other member stations via Satellite Communications.

PBS receives program material from a wide variety of sources: WGBH, WNET, WTTW, Southern Educational Communication Association (SECA), WQED and many others, including independent producers such as Louisville Productions, Annenberg/CPB, and American Playhouse. In all, PBS receives programming from over 100 sources each year.

With so many sources of one inch videotape program material it was inevitable that we would have videotape interchange problems. Cary Wight, Director of Technical Operations, has had first hand knowledge of one inch videotape interchangeability. He received a program that was scheduled to air in two days. The usual procedure for National Program Service material is to screen the show prior to airing for technical quality. The screening process consists of evaluation of the following: color balance, pedestal levels, recorded in drop-outs, vertical blanking, horizontal blanking, front porch, horizontal synchronization, back porch, breezeway, burst width, shedding and drop-outs. During the evaluation, Cary noticed visual impairments that were not recorded in. He could not see this impairment in the demod mode, so he knew it was not recorded in. He tried the tape on different machines with varied results. He was able to get another copy from the producer but it alerted us to the fact that we had some in-house interchange problems.

While investigating our in-house interchange problems we discovered that tapes that were recorded in-house would not properly play back on all of our tape machines. Some of the questions that came to mind while seeking a solution to our interchange problems were:

1. How do we start a very aggressive preventive maintenance program? (Vigilant maintenance solves one-inch videotape interchange problems).

2. How do we get tape operators to recognize possible track straightness problems using waveform monitors during the screening process, even though there may be no visual impairments present on the monitor?

3. How do we stop the prolific addition to the tape library of tapes that will surely cause an air discrepancy in the future?

4. Is it possible for us to measure track straightness to ensure that videotapes conform to the SMPTE Type C standard, (American National Standards Institute C 98.19 M)? ANSI C98.19M specifies:

Record Locations and Dimensions

Dimensions	Inches Minimum	Inches Maximum
Video track lower edge	0.15197	0.15433
Video track upper edge	0.88012	0.88484
Video and sync track width	0.00492	0.00531
Video track pitch	0.007177 ref	
Video track length	16.17181 ref	
Sync track length	1.00866	1.04016
Track angle	$2^{\circ}34'$ ref	

The type of preventive maintenance program choosen is a tier system. Level 1 is designed so that every tape machine can receive this magnitude of maintenance at least every two days. See Table 1: Line Routines, Level 1.

LINE ROUTINES, LEVEL 1

	COMMENT	VT#	HOURS
TRANSPORT FUNCTIONS			
1. Clean and Inspect			
2. Tension Arms, Idlers, Scanner, Guide Block, Tracking.			
3. Audio Stack, Latches and Hinges.			
AIR SYSTEM			
4. Check Filters, Pressure and Air Flow.			
MONITORING			
5. Check Lamps, LED's, Menu Operation.			
6. Check Switch Functions, Scope Intensity.			
RECORD CAPABILITY			
7. Audio/Video Levels - Auto EQ.			
8. Assemble and Insert Function.			
PLAYBACK CAPABILITY			
9. Audio/Video Levels, Control Track Phase.			
10. RF Level, AST Operation.			
11. TBC Operation, Horizontal Phase and Routing Switcher Input.			

TABLE 1.

Level 2 should be accomplished every 500 head hours or sooner. See Table 2: Level 2.

LEVEL 2

	COMMENT	VT#	HOURS
1. Cleaning Capstan Hub.			
2. Parking Brake.			
3. Capstan Shaft.			
4. Vertical Blanking.			
5. Horizontal Centering.			
6. Modulator Frequency.			
7. Mod/Demod Levels.			
8. Audio R/P Levels.			
9. Optimize Video/Sync.			
10. Check Flying Erase.			
11. Good Time Play.			
12. TBC Levels/Ramp.			
13. Reel Motors/Brushes.			
14. Scanner Motors and Brushes.			

TABLE 2.

Level 3 should take place between 800 to 1000 head hours. See Table 3: Level 3.

LEVEL 3

	COMMENT	VT#	HOURS
1. Clean Capstan Motors.			
2. Pin-Scanner Gap.			
3. Tip Projection.			
4. Scanner Tach.			
5. Control Track Phase.			
6. Differential Gain.			
7. Differential Phase.			
8. Audio Frequency Response.			
9. Audio Bias/Erase.			
10. Audio Record Equalization.			
11. Audio Edit Timing.			
12. Guide Block/Pins/Heads.			
13. Pin to Scanner Gap.			
14. 1000 Hours: Capstan Motors/Brushes.			

TABLE 3.

Now whenever we receive a tape that has a suspected track straightness problem, we assume that our machines are causing the problem. In that way we force ourselves to check the videotape machines that are playing back the tape using our Maintenance Technical Evaluation Notice. Only after the playback machine has been thoroughly checked do we concentrate on the producer's tape. By assuming that we are at fault and thoroughly checking the playback machine we hope to avoid the situation where we might think that the producer's tape is at fault when actually our machines are the culprit.

MAINTENANCE TECHNICAL EVALUATION NOTICE

PROGRAM SERIES______ STEREO ______ BACKUP______

EPISODE______ DOLBY A______ SCOTCH______

PRODUCTION CENTER______ CLOSED CAP______ AMPEX______

PBS CONTROL #______ OPEN CAP______ SONY______

VTR #______ MODEL______ DATE______ MASTER______ FUJI______ OTHER______

Guide Alignment Tape -AST Head On- Unity Tracking Playback

1. RF Envelope Shape ______ 2. AST Head Response ______

Record Head Playback Manual Tracking

1. RF Envelope Shape ______ 2. Entrance Guide ______
3. Exit Guide ______ 4. Control Track ID ______
5. Control Track Phase______

Audio Alignment Tape Unity Position Playback All Tracks

1. A-1 ______ VU 2. A-2 ______ VU
3. A-3 ______ VU

Work Tape Record Mode Bars/Tone Unity Position All Controls

1. Demod Level ______ 2. TBC Level ______
3. Tape Preset Type ______ 4. Left Channel ______ VU
5. Right Channel ______ VU 6. INT. Time Code ______ VU

Work Tape Playback Mode Unity Position All Controls

1. Demod Level ______ 2. TBC Level ______
3. Left Channel ______ VU 4. Right Channel ______ VU
5. Time Code PBK ______ VU

Transport Functions

1. Tape Path ______ 2. Air System ___ / ___ / ___ / ___
3. Supply Tension ______ 4. Take Up Tension ______
5. Reverse Shuttle ______ 6. Forward Shuttle ______

Tape Interchange

1. RF Envelope ______ AST 2. RF Envelope ______ REC.HD.
2. TBC Video Out ______

Air Tape

1. Type of Impairment ______ 3. Comment ______
2. Nature of Complaint ______ 4. Evaluator ______

Date ______

The bimorphic Automatic Scan Tracking (AST) head uses the R.F. envelope of the video signal to determine if the head is off track. So we monitored the AST servo error detector. We picked a convenient test point and observed the signal as we mistracked the tape by adjusting the entrance and exit guides. When properly aligned, the AST error voltage is quite small. As the entrance and exit guides were raised and lowered, the AST error voltage increased drastically. We also recorded the results of the detected R.F. (Figures 1-20.)

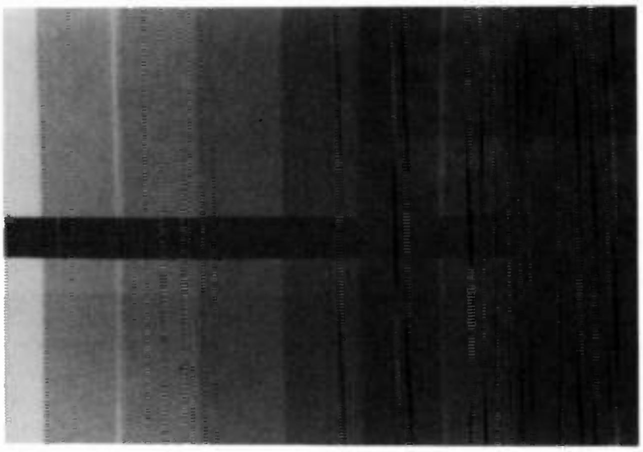

Figure 1: MONITOR NORMAL TRACKING

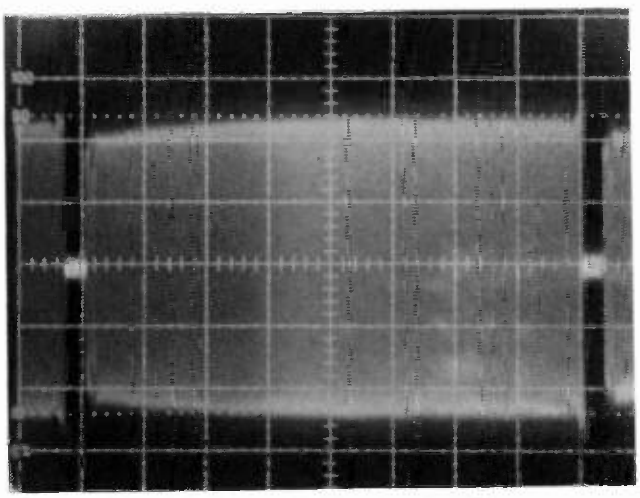

Figure 2: WAVEFORM MONITOR NORMAL TRACKING

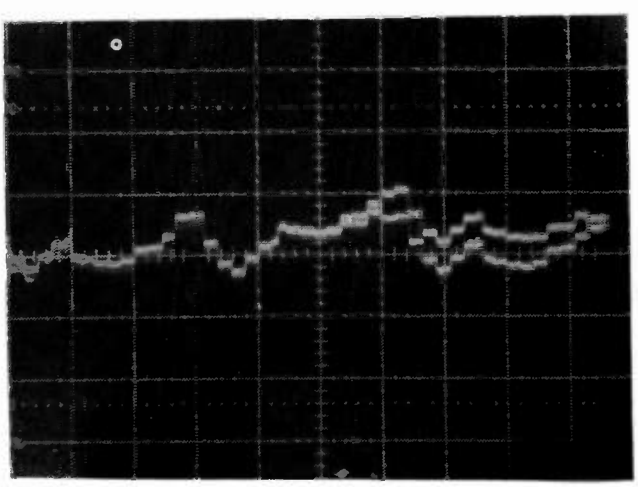

Figure 3: AST ERROR VOLTAGE-TRACKING NORMAL

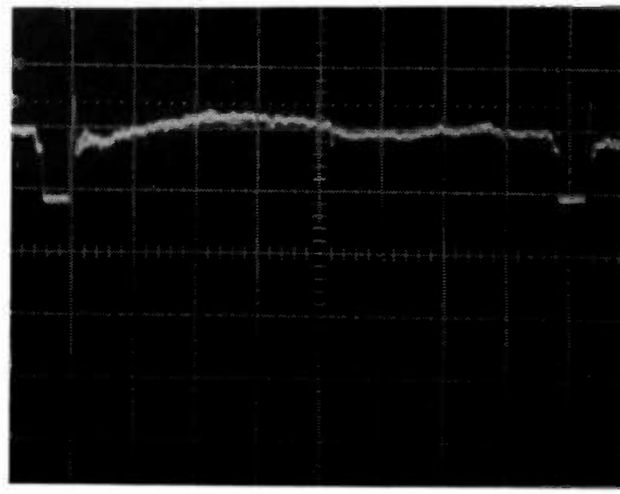

Figure 4: DETECTED R.F. TRACKING NORMAL

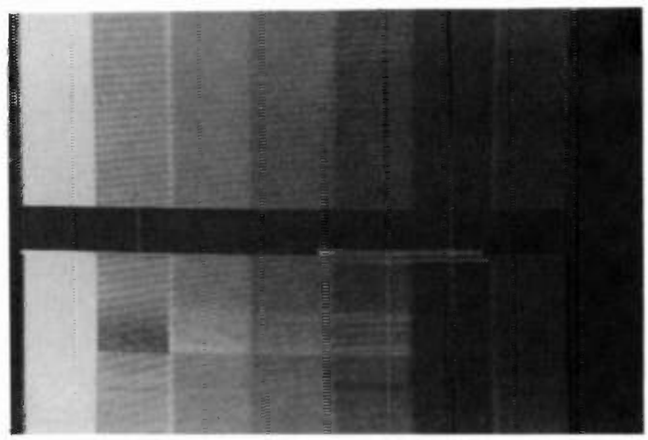

Figure 5: EXIT GUIDE ADJUSTED LOW

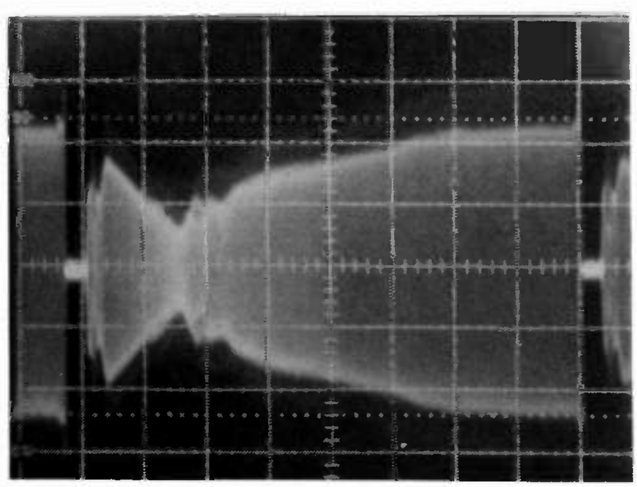

Figure 6: WAVEFORM MONITOR - EXIT GUIDE ADJUSTED LOW

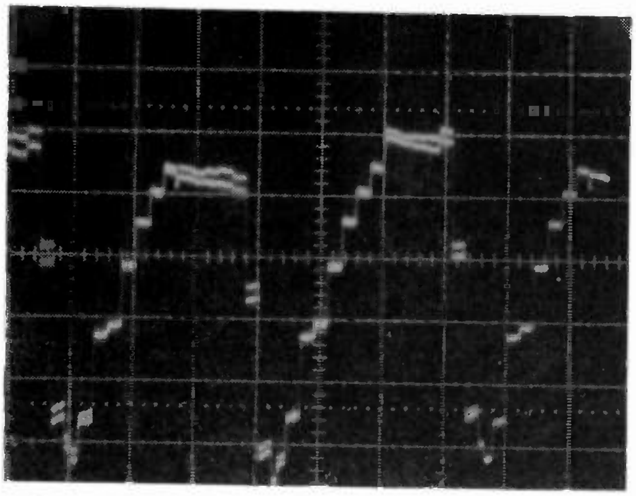

Figure 7: AST ERROR VOLTAGE - EXIT GUIDE ADJUSTED LOW

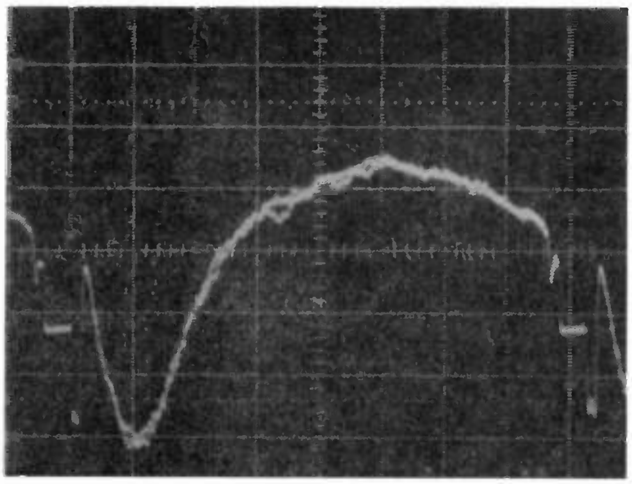

Figure 8: DETECTED R.F. - EXIT GUIDE ADJUSTED LOW

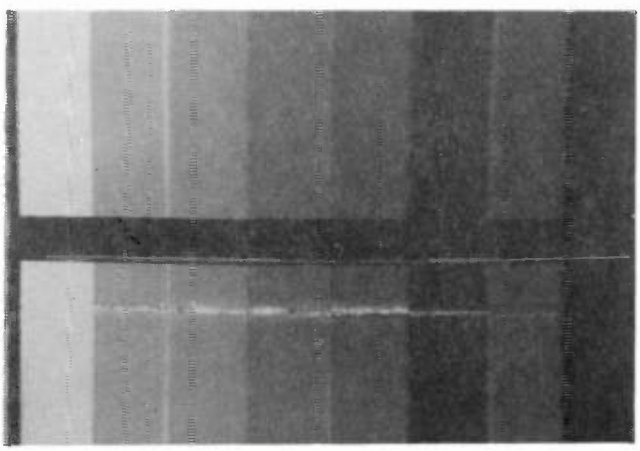

Figure 9: EXIT GUIDE ADJUSTED HIGH

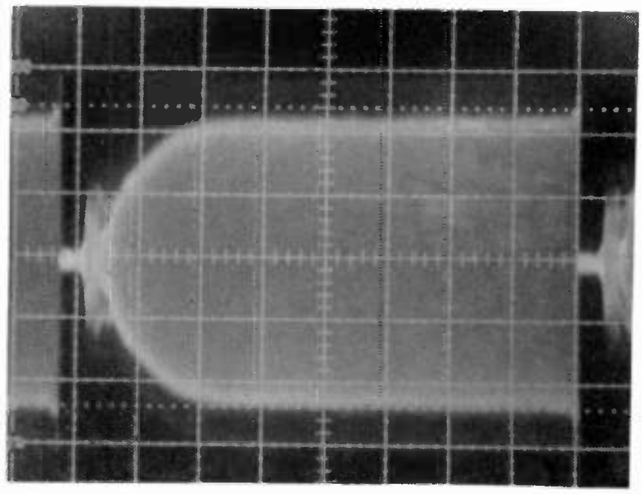

Figure 10: WAVEFORM MONITOR - EXIT GUIDE ADJUSTED HIGH

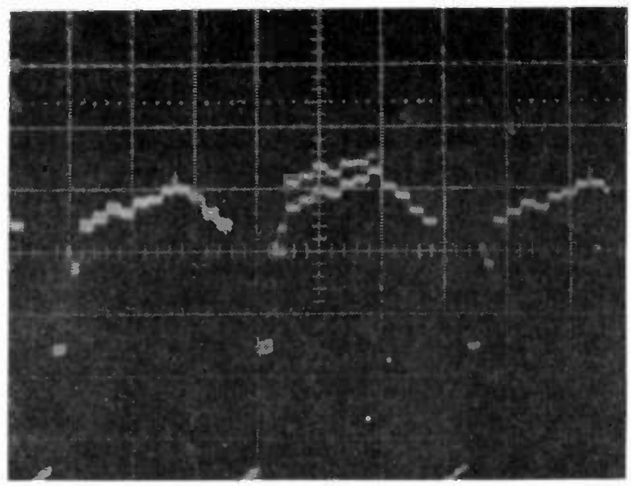

Figure 11: AST ERROR VOLTAGE - EXIT GUIDE ADJUSTED HIGH

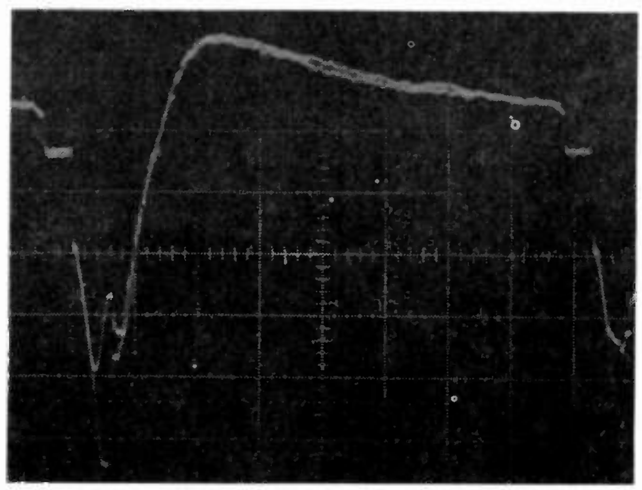

Figure 12: DETECTED R.F. - EXIT GUIDE ADJUSTED HIGH

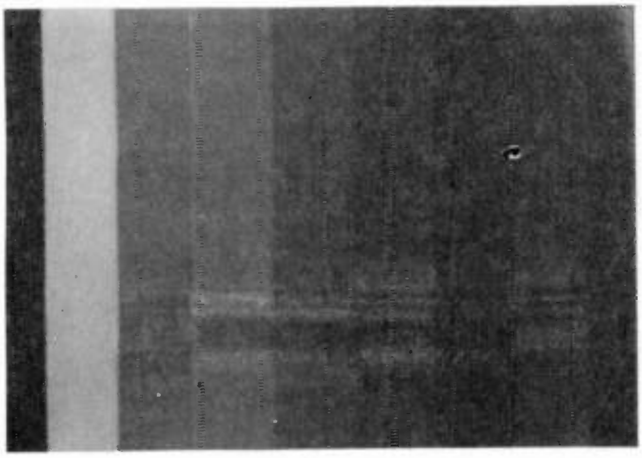

Figure 13: ENTRANCE GUIDE ADJUSTED LOW

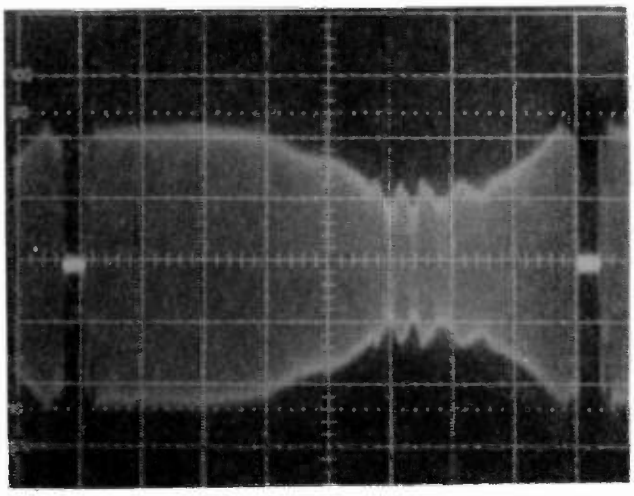

Figure 14: WAVEFORM MONITOR - ENTRANCE GUIDE ADJUSTED LOW

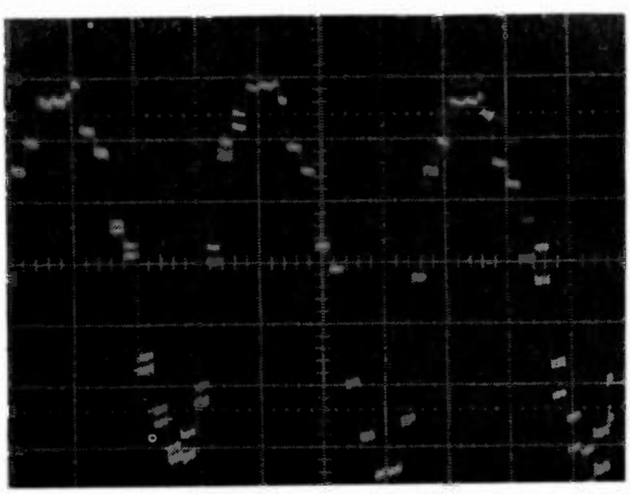

Figure 15: AST ERROR VOLTAGE - ENTRANCE GUIDE ADJUSTED LOW

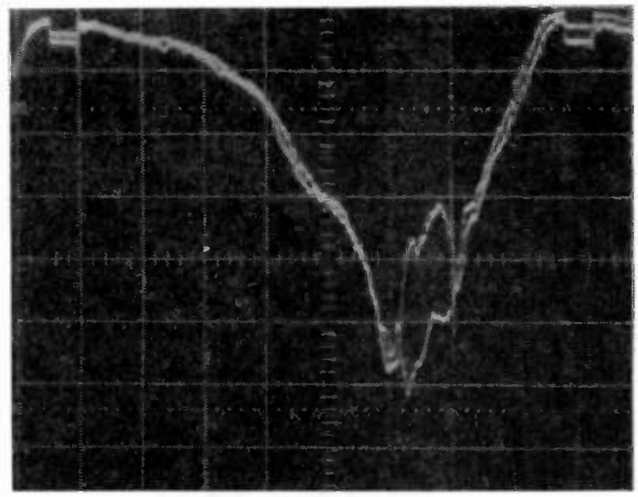

Figure 16: DETECTED R.F. - ENTRANCE GUIDE ADJUSTED LOW

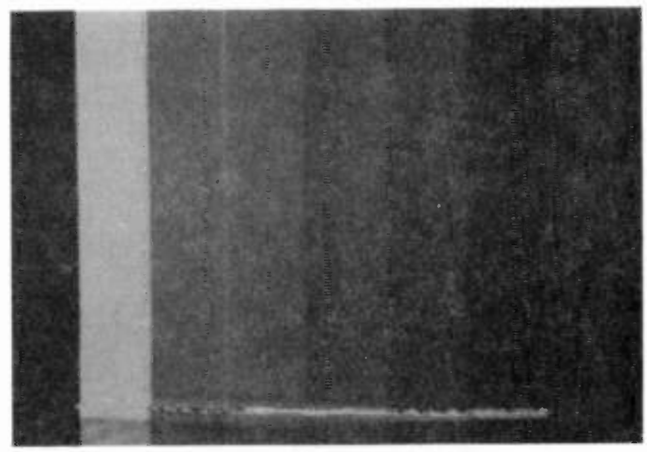

Figure 17: ENTRANCE GUIDE ADJUSTED HIGH

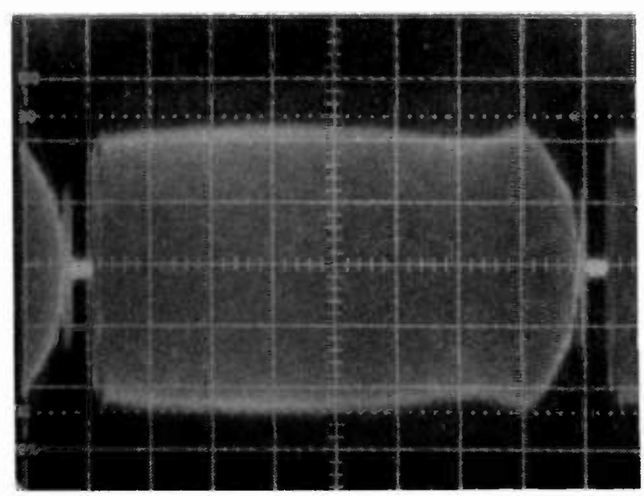

Figure 18: WAVEFORM MONITOR - ENTRANCE GUIDE ADJUSTED HIGH

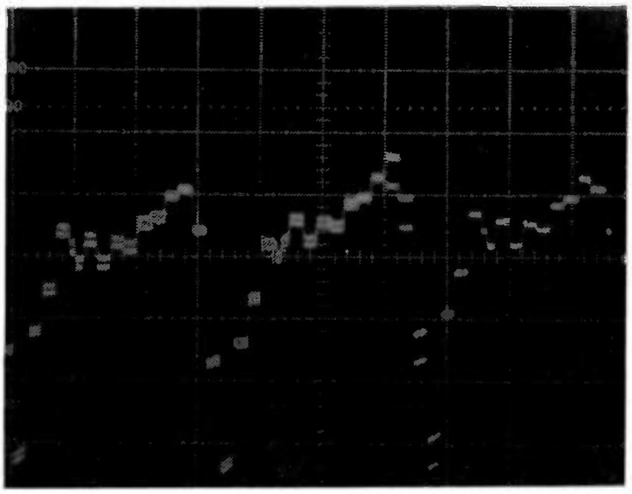

Figure 19: AST ERROR VOLTAGE - ENTRANCE GUIDE ADJUSTED HIGH

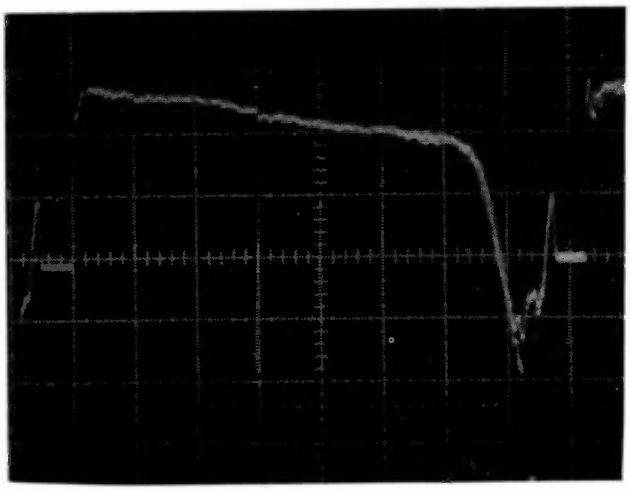

Figure 20: DETECTED R.F. - ENTRANCE GUIDE ADJUSTED HIGH

Figure 21 shows the AST error voltage of five different tape machines. A master tape was recorded with the entrance guide high and then played back on different tape machines displaying AST error voltage. Care was taken to play the same area of tape back on all the machines. While error voltage can be detected on all the tape machines, the level of error varied with each tape machine depending on head wear. This is adequate enough to alert an operator that there may be a track straightness problem, but in no way can be used to reject a tape because the waveforms are so head dependent.

In order to accurately measure track straightness, we adopted a method developed by Ampex Corporation. We actually cut a section of tape off the reel, about nine inches. With a mixture of water-based ferrofluid (#508) we were able to render visible the record locations and dimensions.

With Ampex #196 videotape we found the best mixture of ferrofluid was two parts water and one part ferrofluid. The best method we found of evenly applying the ferrofluild to the tape was saturate a cotton swab wrapped with cloth, and pull the tape over the cotton swab, not the swab over the tape. Allow the tape to dry for 30 minutes.

Using a stereomicroscope with a maximum magnification of 64 (see Figure 22), we were about to clearly see audio tracks 1, 2, and 3, as well as video, control, sync and color frame pulses (see Figures 23-27).

There are 101 video tracks recorded on SMPTE type C videotape. According to ANSI C98.19M - 1979, the video track pitch

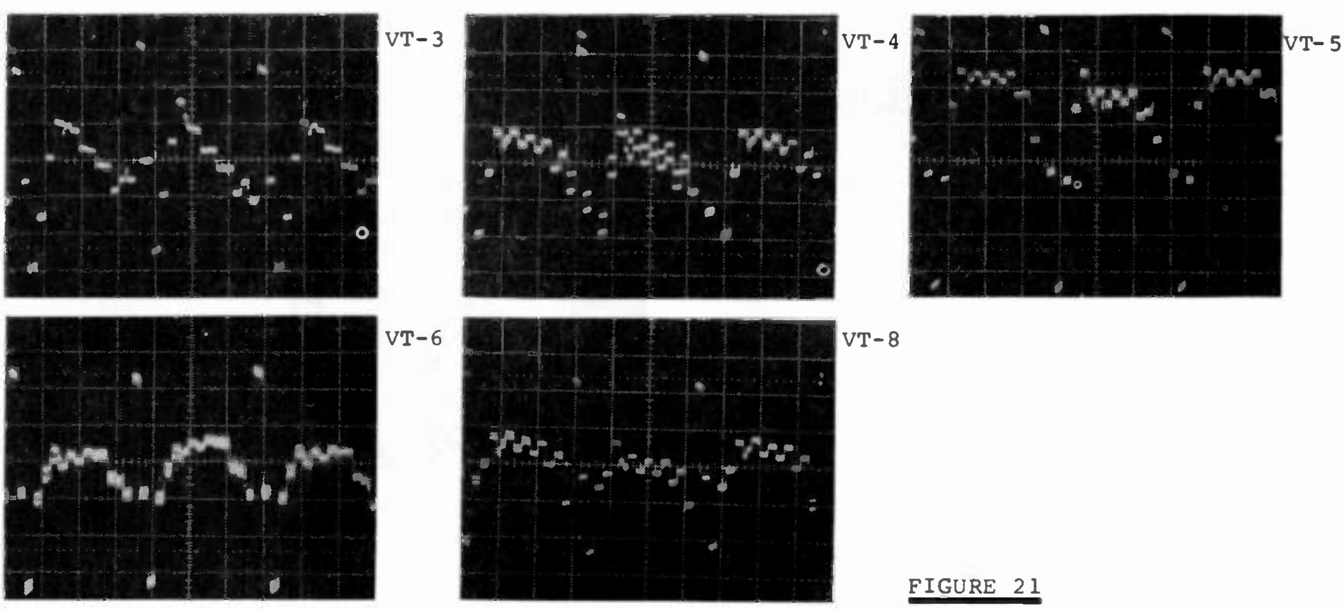

FIGURE 21

(distance between tracks) will be 0.007177 inches. It is not necessary to measure the entire 16.17 inches of video track, nor do you have to measure all of the tracks to determine track straightness. Instead, measure every second track for the first and last ten tracks. Then every tenth track is measured between tracks 10 and 90.

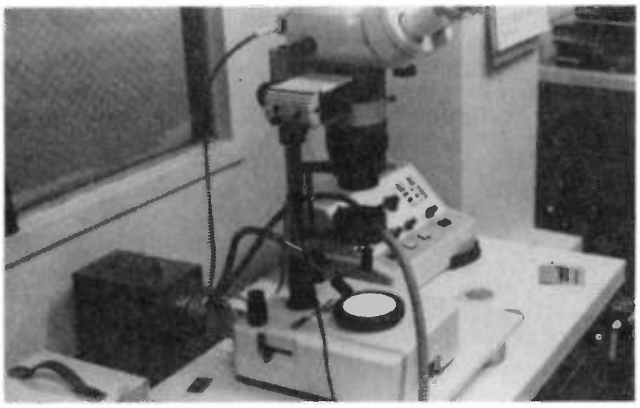

Figure 22: STEREOMICROSCOPE

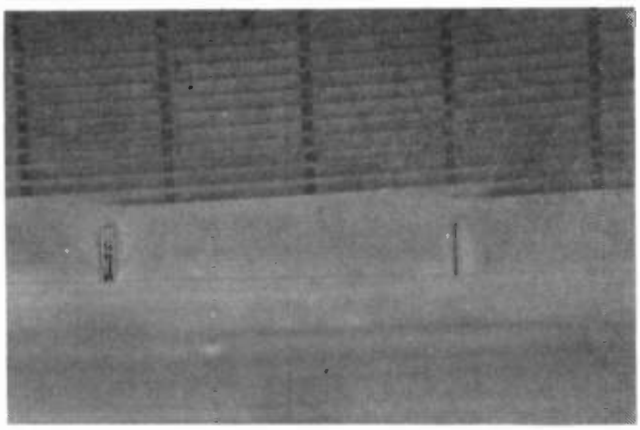

Figure 23: PULSE AT LEFT IS COLOR FRAME PULSE. PULSE AT RIGHT IS CONTROL PULSE. MAG. 12.6X

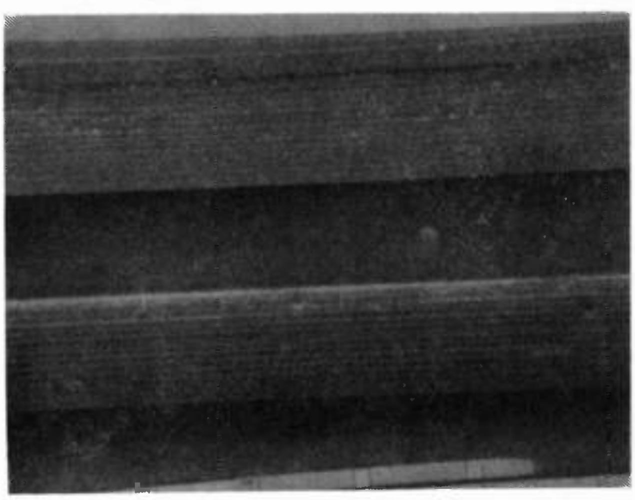

Figure 24: AUDIO TRACK 2 AT TOP, AUDIO TRACK 1 AT BOTTOM. MAG. 20X

To compensate for tape slitting, several readings may be taken along the tape and resemble are used. The readings are taken from edge of tape to the beginning of the track along the Y-axis. Measurements are taken using a calibrated eyepiece or a digital length measuring set.

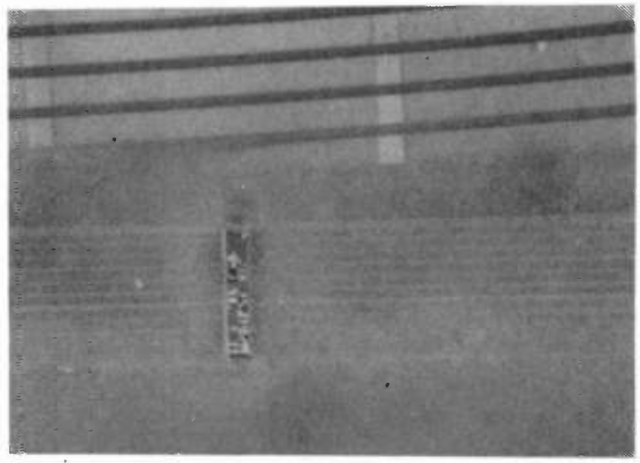

Figure 25: VIDEO TRACKS AT TOP, CONTROL TRACKS AT BOTTOM. MAG. 32X

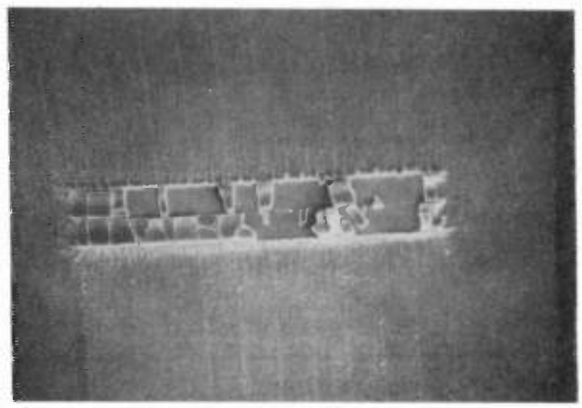

Figure 26: CONTROL TRACK PULSE EXPANDING USING A SCANNING ELECTRON MICROSCOPE. MAG. 130X

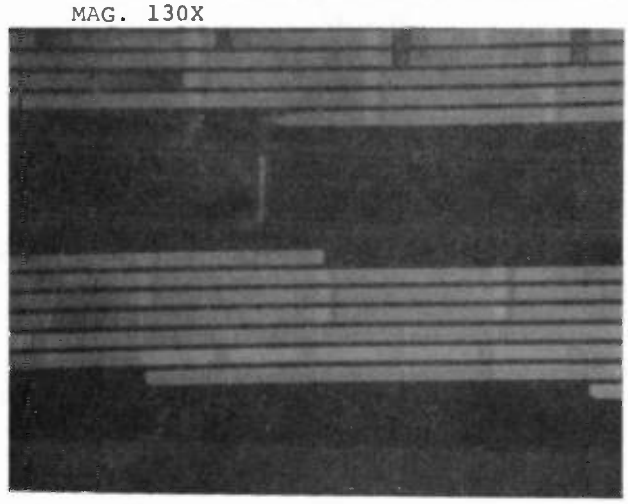

Figure 27: TOP IS VIDEO TRACKS, BOTTOM IS SYNC TRACKS. MAG. 16X

The nineteen readings are entered into a HP-41 programmable calculator, which is running a track error program that derives track spacing errors using least squares * curve fitting technique to calculator linear approximation.

Prior to our fire on Oct 15, 1984, we had 10 Ampex AVR-1's, 8 Ampex VPR-2B's and 6 VPR-2's. Since the fire we have 1 Ampex AVR-2, 8 VPR-3's, 9 VPR-80's and 3 Sony BVH-2000. Our immediate future requirements will be 22 Ampex VPR-3's and 3 Sony BVH-2000's.

During the past two years we have accumulated 91,000 of head hours on 17 tape machines. We have not been plagued with any in-house SMPTE type-C videotape interchangeability since we started our maintenance program. The key to good interchangeability is a clean tape path, proper tension, properly aligned guides and constant preventive maintenance. On most VTR's all the video signals shown can be seen by operator from the bridge of the machine except Detected R.F. Therefore an operator can flag a tape that may have interchangeability problems, for further analysis. We can also measure track straightness on any of our machines to make sure they comply with SMPTE type-C specification.

I would like to thank the following for their special contribution to this paper.

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Cary Wight, PBS, Washington D.C.

Ampex Corporation

Sony Corporation

*REF. Algorithm for RPN calculators
John A. Ball

SOLID STATE IMAGE SENSORS FOR BROADCAST CAMERAS

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1. INTRODUCTION

In electronic imaging, the camera tubes are gradually being replaced by solid-state image sensors. These image sensors have many advantages, such as no burn-in, no lag, geometric precision independent of electromagnetic disturbances or scene contrast, a resolution that is independent of scene movement, low power-supply voltages and low power consumption, small size and low weight. A present state-of-the-art broadcast camera, equipped with solid-state image sensors of the frame-transfer type, is described elsewhere in these conference proceedings (ref. 1) and is exhibited at this conference.

A frame-transfer sensor is one of the possible types of solid-state area imagers. The excellent applicability of the frame-transfer type in present and future camera systems will be shown in this paper. For this application different types of imagers will be discussed, emphasizing a comparison between imagers of the frame-transfer type and the interline type.

2. OPERATION PRINCIPLES

2.1 Light detection

All types of image sensors convert incoming light into charge packets, separately stored in different photosensitive elements. The differences in amounts of generated charge reflect the contrast of the projected scene information. This distribution is collected over some integration period.

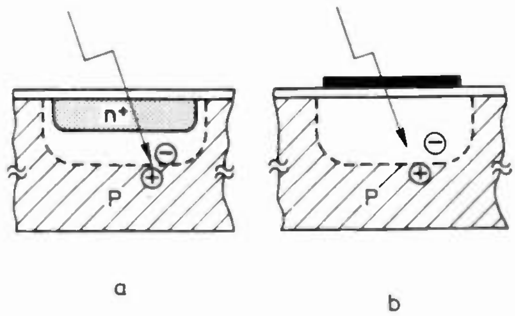

Fig. 1 Main types of photosensitive elements, here in p-type substrate. (a) An n+p diode, (b) An MOS capacitor.

Figure 1 shows two commonly used types of photosensitive elements: the n+p diode and the MOS capacitor.

These elements are normally made in silicon. There, photons with a wavelength shorter than 1000 nm create one electron-hole pair. In the photocell the electron and the hole are separated by the electric fields present in a depletion region. This region is created by the application of a positive bias on either the n+ region or the MOS gate. As a result, the holes are drained off in the p-substrate and the electrons are collected on the capacitance formed by the diode or the MOS gate.

2.2 Charge readout

The mobile charge is finally converted into a voltage change proportional to the amount of charge. This voltage change is produced by forcing the charge packets onto a capacitor which is

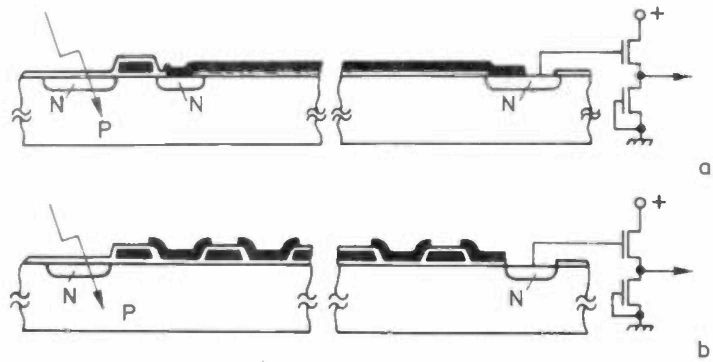

Fig. 2 Main types of charge readout between photosensitive element and output amplifier: (a) MOST switch and sense line, (b) charge-coupled device.

connected to an output amplifier. Two methods are used to achieve this (see figure 2): In the first method a MOS transistor connects the picture element (pixel) with a long sense line, the other end of which is formed by the output node. This sense line has a rather high capacitance, and hence the resulting voltage signal is low.

In the second method the charge packet is transferred from the pixel to the output node by a charge-coupled device. At the other end of this analog shift register the charge is dumped onto a small capacitance which results in a much higher output voltage than in the first method.

3. CONFIGURATIONS OF AREA IMAGERS

The various configurations of area imagers differ in the way that types of photocells and charge-read elements are combined. In this section the three most common configurations are briefly described, i.e. the XY sensor (ref. 2,3), the Interline (IL) sensor (ref. 4-6) and the Frame-Transfer (FT) sensor (ref. 7,8). Each of these sensors contains many thousands of photocells, regularly arranged in a 2-dimensional array. Of course, individual sensors of these three types can deviate from the description given below, e.g. where in the description

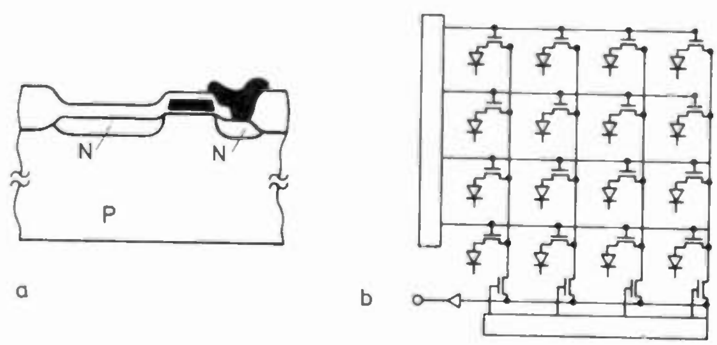

Fig. 3 The XY-addressed image sensor, (a) cross-section through one picture element, (b) diagram of organisation.

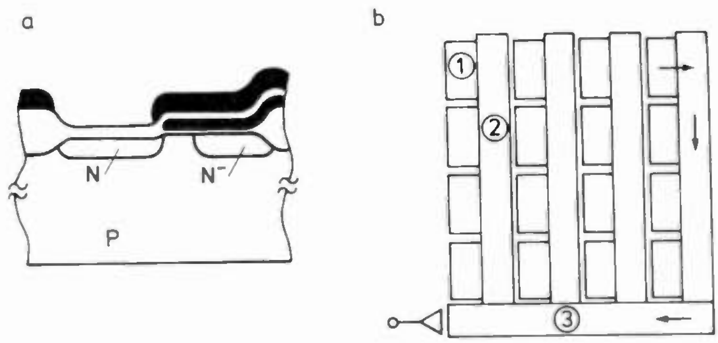

Fig. 4 The interline-transfer image sensor, (a) cross-section through one picture element, (b) diagram of organisation. The numerals refer to: 1. photo diode, 2. vertical CCD, 3. horizontal CCD.

of the IL sensor we assume the photocells to be diodes (ref. 6), they can also be MOS gates (ref. 9).

Figure 3 shows an example of an XY sensor. One pixel contains a photocell and a MOST switch. The MOST switches on one line are interconnected, and addressable via a vertical shift register. A positive bias on such a line connects all photocells on that line to their vertical sense line. With aid of a horizontal shift register each individual vertical sense line can be connected to a horizontal sense line via a MOST switch.

Figure 4 shows an example of an interline sensor. The vertical sense lines are now replaced by CCD shift registers. After an integration time the charge packets in the photodiodes are transferred into the light-shielded vertical shift registers. During the next integration time the charge packets in these registers are shifted line by line into a horizontal register where they are quickly shifted to the output node. In the meantime the photocells can collect the charge packets for a new image.

Figure 5 shows the frame-transfer image sensor. It consists of a large number of parallel vertical CCD shift registers debouching into a horizontal CCD shift register. The upper part of the

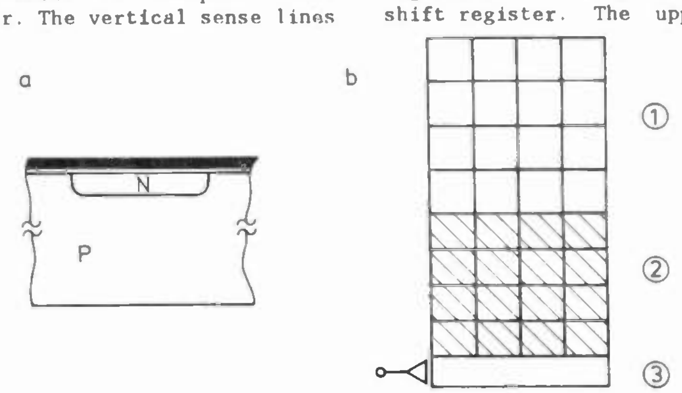

Fig. 5 The frame-transfer image sensor, (a) cross-section through one picture element, (b) diagram of organisation. The numerals refer to: 1. image section, 2. storage section, 3. horizontal readout register.

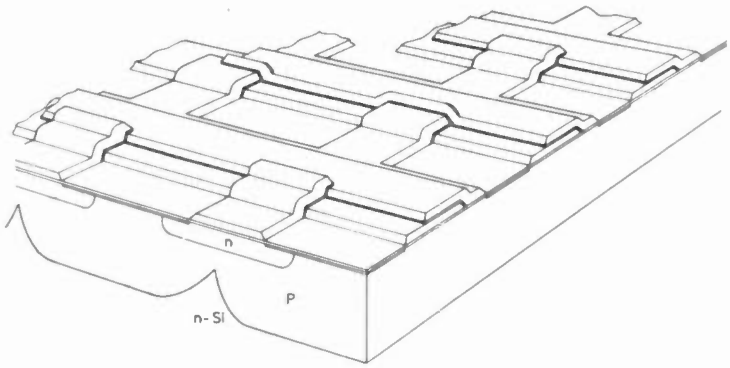

Fig. 6 Perspective of a part of the image section showing the electrode structure, gate oxide, and dopant structure of an image cell. The p-dopant concentration below the n-type transport channels is locally diminished by implanting the p-dopant through stripes parallel to the n-channels. The blue sensitivity of a pixel is increased by not covering parts of the cell with electrodes.

vertical shift registers constitutes the photosensitive image section and the lower part the lightshielded storage section. In the image section each element of a vertical register is a photocell and readout element at the same time. After an integration time all charge packets are transferred from the image section to the storage section. While in the next integration time a new image is formed in the image section, the storage section is emptied line by line into the horizontal register like in the case of the IL sensor.

The main advantage of the XY sensor is its similarity to a MOS Random Access Memory. This allows the direct use of VLSI technologies for RAM's for the production of this type of sensor. A main problem left is its poor sensitivity caused by the large value of the sense-line capacitance. Allthough progress has been made on this point (ref. 3,10,11), the discussion will be continued with FT versus IL only.

4. FRAME TRANSFER VERSUS INTERLINE

In this section we will discuss the FT versus IL, addressing the main aspects which determine their performance.

4.1 Antiblooming and smear

If a pixel is illuminated so intensely that more electrons are generated than can be stored, the excess electrons have to be removed before they can spread into neigbouring cells ("antiblooming"). Nowadays in IL and FT imagers this is mostly done be using the "vertical" antiblooming method (in this context "vertical" means downwards into the silicon substrate). This method can be used if the n+p diode or the CCD channel is built in a p-well on a n-type substrate. Figure 6 illustrates this for an FT sensor (ref. 12). The p-well beneath the centre of the n-channels is made sufficiently shallow and lightly doped so that it is depleted. The resulting potential profile in the centre of the pixel is shown in fig. 7. When the

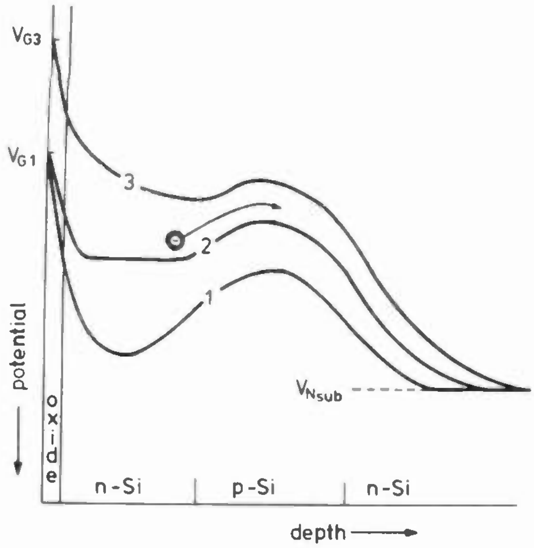

Fig. 7 Resulting potential profiles in the centre of the pixels showing the antiblooming capability. (1) Integrating electrode without mobile charge and (2) full of charge, (3) blocking electrode.

potential well is full (curve 2) the mobile charge in the top-layer "sees" a lower potential barrier to the substrate than to the neigbouring, blocking electrode (curve 3). Hence at overexposure the excess electrons spill over downwards into the n-substrate.

With proper design more than 100x overexposure can easily be handled. However, at such high overexposure a new problem is encountered: smear, a phenomenon which occurs in all types of image sensors. For an FT sensor its origin lies during the shift of charges ("frame shift") from image section to storage section. Figure 8 explains this, assuming that the image section is illuminated only in a small square. During the frame shift the charge packets generated above the illuminated area have to pass this area, thereby receiving additional charges during their passage. The charge packets below the square have already passed the illuminated area during the previous frame shift. In the displayed image this smear effect becomes visible as a less dark region above and below the square.

In an XY- or IL sensor smear manifests itself in the same way, allthough its origin is different. In IL sensors the vertical registers are shielded against light. Nevertheless electrons leak into those registers, e.g. generated directly by oblique incoming light or after many reflections of the incoming light underneath the lightshield. Allthough in IL sensors only a fraction of the light contributes to the smear signal, its effect may be large since it can take an entire integration time before the charge packets reach the perfectly lightshielded horizontal registers.

As will be discussed in section 4.3, the smear signal can become the limiting factor in achieveable SNR. In IL sensors there is still progress in suppressing the smear signal, but it can not be eliminated completely. In FT imagers the smear signal is really zero with the application of a shutter during the frameshift, as is e.g. done in the camera system of ref. 1. Since in IL sensors the smear signal is formed during the total integration time, the use of a shutter to prevent smear is not applicable.

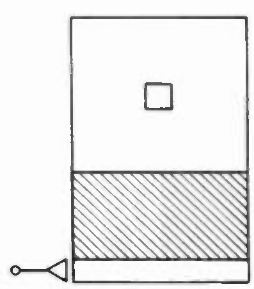

Fig. 8 The origin of smear in an FT sensor.

4.2 Quantum efficiency

The quantum efficiency (QE), i.e. the percentage of photons that are converted into mobile charge, depends on the layout of the photocells and the structure of the layers above the silicon (see e.g. ref. 13). Both for IL and FT sensors the structure of the image cells is still under development and the QE is still improving. For both sensor types the QE for the visible spectrum typically lies between 10% and 50%. For the IL sensor the upper boundary will stay at about 50% because of its concept (light-shielded registers inside the photosensitive area). Allthough the QE of red light can already reach 100% in present FT imagers, it is reduced with aid of a structure as shown in fig. 6 in favour of good antiblooming properties and no loss in resolution (ref. 12). In addition the suppression of red sensitivity makes the spectral sensitivity curve of an FT sensor more resembling the eye sensitivity curve. Blue light is nearly totally absorbed by polysilicon material. To increase blue sensitivity, "windows" are made in the polysilicon electrode structure of FT sensors (see also figure 6). If in the future transparant electrode material becomes available, the QE of FT sensors can approach 100%.

4.3 Dynamic range

The dynamic range in a displayed scene (excluding shot noise) is defined as the maximum sensor signal divided by the sensor signal in darkness. For an FT sensor the maximum signal is determined by the maximum well capacity in the image section, roughly 150,000 electrons. With proper signal processing the noise in darkness is determined by the fixed-pattern noise only. Typical values for this noise expressed in electrons per pixel are 30 at room temperature and 200 at 60 degrees Celsius. This leads to a dynamic range of 74 dB at room temperature, reducing to 52 dB if shot noise is included. These values are typical for present state-of-the-art FT and IL sensors.

The contrast in the scene may exceed this dynamic range without disturbing the image. As the number of times of overexposure is increased the smear signal will eventually limit the dynamic range. As discussed before, in an FT sensor smear can be totally eliminated with a shutter. Therefore for an FT sensor the contrast in the scene may be much larger than in IL sensors before degrading the dynamic range.

4.4 Resolution and Moire

Because solid-state imagers take discrete samples of the image, Moire or aliasing effects play an important role. When the spatial frequencies in the scene exceed the half of the sampling frequency, false low-frequency components are found in the response. Two parameters are of importance: the width of the photosensitive part (aperture) of a pixel and the pixel pitch. For a given optical format always a balance has to be found between increased resolution and reduced Moire effects. In IL sensors the small (horizontal) aperture of a pixel allows small details to be resolved, but the associated higher MTF increases aliasing which then has to be tackled by optical low-pass filters. In our view pixels which are photosensitive over their total width (as in FT imagers) offer a very good compromise between resolution and Moire.

However, the main measure for achievable resolution remains the total number of pixels. Optimization of pixel design leads to second order improvement. For the definition in horizontal direction, an FT sensor needs only the space of two lithographic details, whereas an IL sensor needs four. In vertical direction both the IL and the FT sensor need two lithographic details. For the FT sensor the latter is the result of recent progress with the introduction of an accordion imager (ref. 14).

The accordion concept is explained in fig. 9. showing the potential wells along a vertical column in the image and the storage section. For simplicity we assume

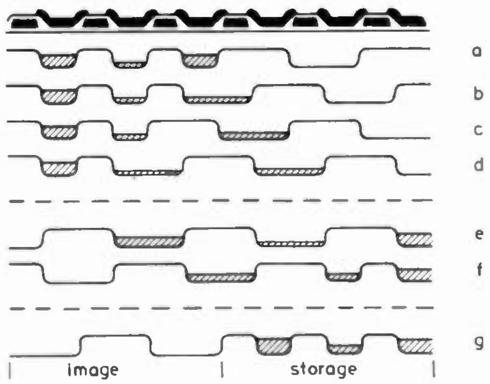

Fig. 9 The accordion readout mechanism along a vertical column: starting from two-phase integration in the image section, going to 4-phase transport mode and ending with close packing in the storage section.

that both sections contain three cells only. During integration half the electrodes in the image section are positive (collecting charge), the others are at zero bias (blocking). Just before the frame shift starts (fig. 9a), the storage section is empty and its electrodes are clocked in a four-phase mode. By applying the proper voltages to the overlying electrodes, the charge packets nearest to the storage section start to move (fig. 9b). As soon as there are two blocking electrodes between this first packet and the next (fig. 9c), this next charge packet starts to move (fig. 9d). This process continues until all charge packets are moving. At that moment the first packet has just arrived under the last two electrodes at the bottom of the storage section (fig. 9e). As soon as there is only one blocking electrode between this first packet and the second, the next electrodes stop clocking (fig. 9f). The sequence stops when all packets are closely packed in the storage section. The readout of the storage section into the horizontal registers in the next integration time occurs by a similar process. In the accordion imager the electrodes are driven by on-chip digital CMOS shift registers which produce the clocking sequence.

5. CONCLUDING REMARKS

As seen above the FT sensor has the capability for a two times better horizontal resolution than the IL sensor, when the optical format and the minimum design rules are the same. With this smaller pixel size, the FT sensor can offer the same dynamic range as the IL sensor. The FT sensor has the flexibility to exchange the potentially high resolution against even better spectral sensitivity by enlarging the pixel width. The price one has to pay for the FT

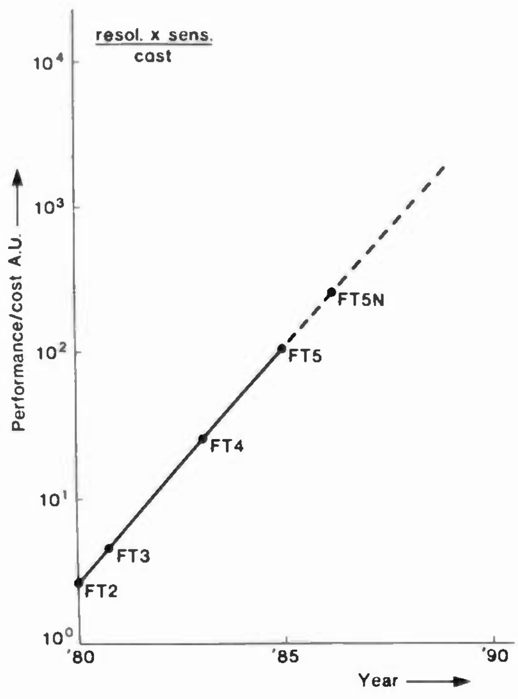

Fig. 10 Progress in solid-state imaging: the figure of merit of consecutive frame-transfer imagers is plotted versus the year of their design. (More details: FT2 and FT3: ref. 8, FT4: ref. 12, FT5: ref. 14).

concept are its larger chipsize (because of its storage section outside the photosensitive area) and the application of a shutter which eliminates the smear completely, thus offering a high dynamic range even under intense illumination.

Finally the progress in solid-state imaging is illustrated with the aid of figure 10. This figure shows the performance divided by cost per chip versus time. The datapoints refer to several types of FT sensors developed in our laboratories. This figure shows three orders of magnitude improvement in one decade. We are convinced that the relative youth of solid-state imaging and the continuing impetus of VLSI technology will cause progress to continue at this rate for at least another decade.

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APPLICATION OF FRAME TRANSFER CCD SENSORS IN A NEWLY DESIGNED LIGHTWEIGHT CAMERA SYSTEM

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"The fulfillment of the human wish to have an imaging device as good as his eyes, has come a major step closer, as a result of the progress in CCD imaging sensor technology."

The subject of this paper is the development project undertaken in BTS to utilize advanced frame transfer CCD technology in the design of a lightweight portable television camera system.

The following aspects of the project are highlighted:

- The design goals
- The important role of the camera user during the development
- The use and advantages of a frame transfer CCD
- Expectations for the future

The project started with setting design goals for the product development, including the desire to combine the latest technology with the exacting demands of the camera user for the human interface: See the following Table:

Major Design Goals for the Camera System

1. First as presented above, the excellent ergonomic and operational design.
2. Highest possible picture quality. The quality aspects will be discussed later in relation with camera tubes and different CCD sensor types.
3. Robustness, reliability and no need for maintenance. These are now really possible due to the application of the right type of CCD sensors (frame transfer), which replace the tubes with their ever changing characteristics.
4. Highly automated manufacturing to ensure constant quality, reliability, efficiency and thus attractively priced products.
5. Future proof; this means provisions in the design to enhance product specifications and applications in the future.

1. Excellent Ergonomic and Operational Design

In the field of portable camera design, the constraints for the design engineers were determined by available technology, which always resulted in compromises between performance and ergonomics. The latter being responsible for the "Feel" of the camera and its acceptance by camera people who have to work with it.

In order to achieve the best possible performance and ergonomics, a series of workshops were scheduled to be held coinciding with critical decision stages in the development program and involving the participation of well-known and very experienced cameramen from an international cross section of broadcast television organizations. The workshops provided a unique opportunity to optimize camera ergonomic design in the absence of space consuming tubes, and a design evolved taking maximum advantage of CCD technology. The first workshop took place in February 1985 and was attended by ten "Expert" cameramen from six different countries. It was important to create a free thinking atmosphere where ideas could emerge and be assimulated by the engineers who would be designing the product.

At this one week meeting, several means were available for theoretical and practical discussions about the daily inconveniences with camera and recorder equipment. Three teams were formed, each to design a camera model which was actually made during the workshop. Each team tried to design for the best ergonomical and operational product. After the mock-up was made, it was evaluated by the team itself and by the joint meeting.

The discussions and evaluations resulted in very detailed ergonomic recommendations including: size, weight, balance, length, width, height, unobstructed view over the camera, shape of bottom, viewfinder mounting and adjustment to the individual cameraman.

Other aspects about handling of the camera, on shoulder, tripod, hand held, as well as on motor bike, in car or helicopter were discussed. Of course, the use of accessories such as, microphones, lights, batteries were also reviewed.


DESIGN WORKSHOP

Further detailed recommendations were made for operational controls, viewfinder performance, signaling and indications of functions and controls. Questions were raised for more operational possibilities for more demanding and creative television artwork, all based on single camera and or camera recorder combinations.

A positive experience for all involved parties, was that on most aspects, unanimous opinions were reached. A firm basis for BTS to start the real design and development of the product. However, it was difficult to use and interpret all these inputs correctly, because theory and practice never quite match. For this purpose, a second workshop was organized to verify and to make sure that our interpretation of workshop 1 was correct.

This second workshop was organized 16 months later in June 1986 with nearly the same group of cameramen. At that time, five working try-out cameras in different configurations (single camera, and camera recorders) were presented for evaluation and criticism.

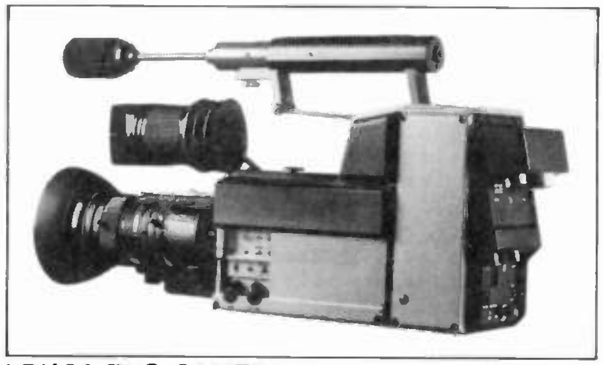
By using the equipment in different teams under operational circumstances, nearly all ergonomical aspects were considered and tested. Now the Frame Transfer CCD camera pictures could be judged in an early stage.

Workshop 2 proved that we were on the right track. Many aspects were reexamined and therefore, many changes and improvements were defined which nearly all could be implemented for the final production models.

Immediately after this workshop, the cameras were demonstrated to eight different broadcast organizations in order to obtain technical and additional operational comments and views from technical, planning and management oriented people.

In all, the workshops proved to be an excellent method to combine the efforts of industry and users, to obtain a marketable product for mutual benefit.

The results of these workshops have been built into the LDK-90 Camera system and this know-how is of course also available for future products.


LDK90 ENG CAMERA

LDK90 CAMERA RECORDER

Ergonomical Benefits to the Camera User:

1. Very low profile, unobstructed view to the right of the camera man.
2. Triangular viewer, x-y-z adjustable and rotatable.
3. Flat camera base with adjustable shoulderpad for perfect balance in ENG, EFP and camera recorder modes.

4. Ultra low optical axis, enables an extremely low shooting position of 70 mm.
5. Compact, lightweight.
6. Integrated battery, low power consumption.
7. Rugged magnesium diecast housing. RFI and rain protected.
8. Easy controls with status indicators in viewer.
9. Field adaptation for ENG/EFP/cameras recorder.
10. Wide range of accessories: lenses, matte box, lights, microphones, painting control panel.

Operational Benefits to the Camera User:

1. No Cap required. No burn-in risk.
2. No color filter wheel required. No loss of sensitivity by color filters.
3. Two electronic presets for color temperature. (studio and daylight)
4. Two auto white balance memories with very wide range.
5. Menu control panel for selection of personalized settings (Gamma, Contour, Black level, Black stretch, and (in preparation) Exposure time).
6. Low gain position for extremely high S/N ratio: Enhances S/N ratio in operational conditions with 6dB.
7. Dynamic contrast compression handles over 500% signal level over the entire white balance range.
8. Contours from red and green.
9. Momentory iris control with spot measurement.
10. Best possible viewfinder picture with framing borderline.

2. Highest Possible Picture Performance

Several criteria can be used to compare or to determine "Picture Quality". Tube cameras and CCD cameras are available now to compare. In many respects, CCD camera pictures are superior:

* registration
* no lag
* no burn-in
* perfect geometry
* no ageing
* no influence from magnetic fields
* no influence from mechanical shocks and acoustic noise (microphonics)
* uniformity of resolution
* no beam pulling

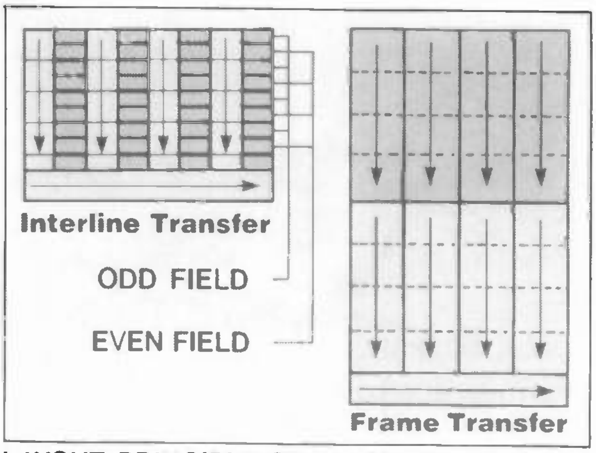
Other criteria are less clear and depend on the type of CCD sensor, for example:

** sensitivity and S/N ratio
** horizontal resolution
* optical format
* integration time (frame or field)
* vertical resolution
** smear
** fixed pattern noise

**Items are extensively discussed in Ref. 1

At present, various CCD sensor types are known, of which two types dominate the market:

* The frame transfer CCD (FT)
* The interline CCD (IL)



LAYOUT PRINCIPLE OF IL AND FT SENSORS

The main differences between these two types are:

- The IL CCD has optical and storage (=transport) areas interleaved on optical surface.
- The FT CCD has separated optical and storage areas. Transport from picture signals from optical to the storage area is done by the optical pixels itself (double function) during the frame blanking time (frame transfer).

Major consequences are:

Sensitivity

- The optical surface of IL-sensors is only partly (for about 35%) sensitive to light.
- The optical surface of FT-sensors is in principle totally sensitive to light.

Horizontal Resolution

Horizontal resolution is for both sensor types depending on the number of optical pixels.

FT sensors will have more (approximately double) optical pixels (= resolution) on the same optical format as IL sensors (using the same minimum design rule).

Optical Format

It will be clear that sensitivity, resolution and optical surface format are closely related to each other. In practice, IL-sensors are mostly 2/3" format, while FT-sensors are of the 1/2" format, with however, higher resolution and sensitivity using smaller optical parts.

Integration Time

- IL sensors have discrete pixels for odd and even fields, which result in an integration time of 33.3 msec NTSC, 40 msec PAL. This equals exposure times of 1/30 respectivly 1/25 sec.
- FT sensors produce odd and even fields with integration times of 16.7 msec NTSC, 20 msec PAL. This equals 1/60 respectively 1/50 sec exposure time.
- The integration (exposure) time influences the dynamic resolution, which is important for moving objects, for example in sports, and should be as short as possible to obtain clean and sharp slow motion and freeze frames.

Vertical Resolution

No significant difference here between IL and FT sensors. Both are better than tubes. However, some IL-sensor cameras can be controlled to simultaneous read-out of odd and even frames; (purpose: integration time 50% less, to obtain better dynamic resolution). This, however, reduces vertical resolution by 50%.

Smear

FT-sensors have a fixed percentage of smear, caused by scene light during the frame transfer (typical: 0.25%).

IL-sensors ideally are free from this phenomena. However, in practice similar "Smear" is produced due to internal flare (spectral wavelength dependent: mostly red) on the sensor surface and caused by insufficient shielding of the transport pixels. This effect becomes worse when the dimensions of the pixels are made smaller for better horizontal resolution. Typical values vary from 0.1 to 0.5%. (These values do not apply to IL-sensors with separate frame store).

Shutter

The smear trouble, however, can be completely prevented by using an optical shutter together with the FT-sensor. The shutter blade shuts incoming light from the sensor during the frame transfer time. This solution cannot be used in combination with IL-sensors.

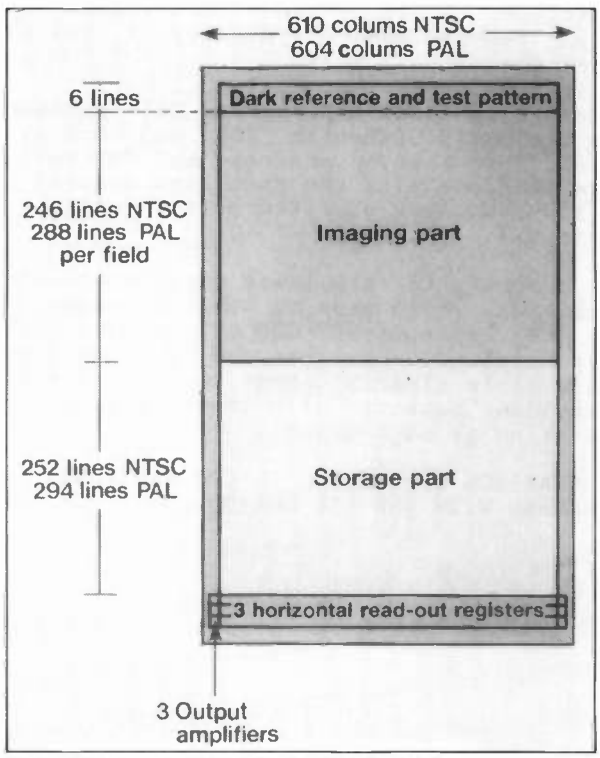
Fixed Pattern Noise

No significant difference between the FT and IL sensors. Fixed pattern noise is temperature dependent and can therefore become troublesome at high temperatures.

Conclusion:

It will be obvious that these differences and possibilities are strongly in favor of the frame transfer sensors. Therefore, our opinion is that CCD-FT sensors are better suited for broadcast applications.

The CCD-FT sensor used in the BTS LDK-90 camera is from the Philips NXA1010/1030 series. Three of these sensors are used with a conventional optical prism beam splitter, as used in tube cameras, and a mechanical shutter. The sensors are bonded to the prism and sealed for the camera life time.



PHILIPS FT-SENSOR FOR LDK 90

Resolution

The high horizontal pixel number (610 NTSC, 604 PAL) ensures high resolution performance: theoretically 64% modulation at 5.8 MHz in the red, green and blue channels. In practice, typical values for red, green and blue of 40% modulation at 5MHz are to be expected which is comparable to 2/3" plumbicon performance. This is a remarkable improvement of about 1MHz bandwidth over typical 500 pixel CCD cameras.

Aliasing

In the FT CCD sensor, the pitch and width ratio of the optical pixels is close to 1:1. Due to this, an optical low pass filter effect is obtained in an electronic way. (Ref. 1). This effect, combined with the high pixel number, contribute to a considerable reduction of moire caused by aliasing.

Sensitivity and S/N Ratio

The 1/2" optical format of the CCD-FT sensor, combined with the highly efficient F1.4 prism, offers a excellent sensitivity of 1750 lux at F4.0 and 90% reflectance, and limiting sensitivity of 27 lux at F1.4 and the highest gain (+ 18dB). It should be noted that all 1/2" lenses have F1.4 aperture, while portable ENG lenses for 2/3" format have typically a maximum aperture of F1.7 only.

The S/N ratio at the normal gain position is typically 56dB with PAL, and 58dB with NTSC. This can be enhanced by 6dB in the low gain position and thus also results in a 6dB reduction of fixed pattern noise (if present).

As a result of all these considerations a table has been made to show a comparison of the BTS-LDK-90 CCD FT camera with typical IL-CCD cameras and tube cameras. This table clearly shows the positive and negative aspects of these cameras in relation to each other:

COMPARISON OF TYPICAL IL-CCD AND TUBE CAMERAS WITH THE BTS LDK-90 CCD FT CAMERA:

PERFORMANCE	BTS LDK-90 CCD FT	TYP 2/3" CCD IL	TYP 2/3" TUBE
LAG, BURN-IN	++	++	--
SMEAR	++	--	++
HOR. RESOLUTION	++	-	++
VERT. RESOLUTION	++	++	-
DYN. RESOLUTION	++	+	-
REGISTRATION	++	++	-
GEOMETRY	++	++	--
ALIASING	+	-	++
POWER CONS.	+	+	-
SENSITIVITY	+	+	-
STABILITY	+	+	-
AGEING/ DEGRADATION	NO	NO	YES
MAINTENANCE	NO	NO	YES
FIXED PATTERN NOISE, (AT HIGH AMBIENT TEMPERATURES ONLY)	-	-	++

++ MUCH BETTER
\+ BETTER
\- LESS
-- WORSE

3. Robust, Reliable, No Maintenance

Having CCD-sensors, the camera is really solid state and can be expected to be maintenance free. To make this true, the built-in shutter has been designed that under normal circumstances, no maintenance or replacement is to be expected during the lifetime of the camera.

The 1.5 gram only shutter blade, with its small maintenance free motor, is dust free, mounted in the sealed optical "Heart" of the camera.

Together with the diecast magnesium camera housing and viewfinder, this product can really be expected to be a robust and reliable tool for todays broadcast use.

4. Automated Manufacturing

The printed circuit boards, a major part of the camera, are automatically assembled and all components are individually tested by means of an ICCT (In Circuit Component Test) measuring system. This will ensure fault free reliable production quality and in addition, efficient, fast manufacturing resulting in competitively priced products.

5. Future Proof

An important function which will become available, is exposure control. This will electronically control the integration time of the optical part of the sensor, resulting in reduced exposure times. (Normal, 1/200 sec, 1/500 sec). This is highly useful for applications, where slow motion and freeze frames will be needed as in sports. The BTS LDK-90 camera has provisions to incorporate this feature.

Because of its high performance, this camera will not only be used for ENG, but also for more demanding applications as documentaries, drama and sports, and will be a perfect partner for the BETA SP and MII format recorders.

Accessories for these applications, as well as an EFP version with Remote Control Unit (RCU), are made part of the LDK 90 camera system.

Based upon the results with the LDK-90, it may be expected that more camera systems with frame transfer sensors will become available in the future from BTS with further improved specifications regarding resolution and sensitivity for all broadcast applications.

Conclusion

CCD technology has so much advanced during the past years, that it has become a viable alternative with many advantages over tube technology.

Frame Transfer CCD's have inherent characteristics which are superior to those of the Interline CCD's; an advantage which will become more important as CCD's are improving.

The use of workshops to aid in determining the needs and desires of the future users has been tremendously successful and has led to a product which we expect to fully satisfy the needs of future camera users.

We would like to thank all participants of the workshops for their contributions.

Ref. 1: M.J.H. Van de Steeg, Philips Research Laboratories, The Netherlands Solid-State Image Sensors for Broadcast cameras (These conference proceedings).

Ref. 2: Dipl. Ing. W.P. Buchwald T.U. Braunschweig - W. Germany

INTEGRATING THE MODERN TELEVISION CART MACHINE WITH TELEVISION STATION AUTOMATION SYSTEMS

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Anaheim, California

INTRODUCTION

The integration of modern cart machines into television stations, where station automation equipment exists or is planned in the near-term future, deserves some careful consideration on the part of those responsible for interfacing these systems to the maximum advantage of the user. This is especially true when one considers a television cart machine with cassette capacities of 250 and above, and with tape recorder complements of between 4 and 6 recorders, or recorder/players.

Since the designers of station automation equipment and cart machines have focused their design efforts toward different objectives, it is important that each take the time to understand the others perspective if the broadcast industry is to be served to the best of our abilities. Examples which highlight the present circumstances include the following considerations:

1. Station automation systems interact with traffic systems to assimilate the daily program schedules. Therefore large video cart machines must also interact with either the automation system or the traffic system to assimilate the daily schedule, so that the cart machines can examine status of their large libraries and print exception reports, thus naming the existing carts that need to be loaded and the non-existing carts that need to be recorded.

2. Most station automation systems were designed with the objective of playing one event for each trigger from an identified source. Cart machines can be expected to play several or even hundreds of sequential events from a single trigger or from the station clock.

3. The process of downloading a full day's log from a traffic system can be time consuming to either the station automation system or the cart machine. Therefore, when the last changes are made, means must be provided to assimilate these changes in both data bases with reloading. In this way, the automation system can execute, track and print "as-run" logs of the operation. Meanwhile, the cart machine, which frequently executes long strings of events, must also interact with the station automation to provide identification of each event executed.

These are but a few of the factors which warrant consideration by the designers of all forms of station automation equipment if their customers are to enjoy the full potential of this emerging technology.

THE AUTOMATION INTERFACE

The TCS2000 represents a large cassette capacity (280) cart machine whose design included an Automation System Peripheral Adaptor which could be interfaced with existing station automation systems using software modules. Similar interfaces, that are supported by several vendors of station automation equipment, already serve older cart machines such as the ACR 25, the Betacart, and others. The concept therefore has been well proven in practice.

Most automation systems are always aware of the physical location of every asset (e.g., a cassette containing a spot). The reasoning here is twofold: the first is to give the automation system sufficient information to perform the library management task; the second rationale covers a system failure. In this case, the operator could be told from a single point (the automation system CRT terminal) the physical location of all material required, so the material could be manually aired.

In resolving the previous two statements regarding operational strategy, the concept of the Table of Contents (TOC) has been developed. A multi-event peripheral (capable of containing several events, such as boxes on the belt of the ACR25 or boxes in the TCS2000 physical library) can be interrogated by the automation system for a list of its physical assets.

The automation interface in the above situation, when interrogated for a TOC data transfer, will provide the following minimum of information:

(a) Physical Location Identifier - This would be in the internal column-row format (i.e., A02 for column A, row 2).

(b) Material Identification Number (MID) - This is the logical "house" number known to the data base. This MID number could identify material physically present in the cart library, as well as material known to the cart machine but not currently present.

(c) Box Number - The 6 digits of the physical Bar Code label attached to every box in the cart machine library could be returned to the automation system if desired.

Additional information such as Title, SOM and Duration could be returned if necessary. However, this information is always available to the cart machine by way of its data base. Due to the large number of boxes the library is capable of containing (281), it is recommended that only required information be sent to minimize communication time and overhead.

Play lists, that may run from one to 24 hours in duration, could be sent from the automation system at any time. Typically, at start of operation for a given day, as much play list as is available would be sent to the cart machine. The TCS2000 library management software could then be commanded by the operator to print an "Exception Report." This report lists those items that are known to the machine but are not physically in residence (events that will need to be loaded). Additionally, this report lists those elements that the cart machine is responsible for but it does not yet know about. These "missing from database" events are ones that must be recorded onto carts using the cart machines' record mode.

The playlist processor, in conjunction with the library and data base, would return to the automation system the status of each requested playlist element. Possible status returned would include:

1. Present in the cart machine's library.

2. Not present in the cart machine, but a known box. An operator would, prior to air time, get and load this box through the cart machine's load port.

3. Unknown Cart Material: This reply assumes a field in the playlist transfer to indicate that this material is expected to be played on a(the) cart machine. The machine flags this material as "not in data base." The action of recording the material in the record mode enters this information into the data base and causes the play list to reflect the correct new status. Alternately, if the material was(is to be) prepared on another system, the information can be directly entered into the machine's data base.

4. Not Cart Material: This status could be returned for all non-cart items if desired.

With the above, and other operational information at their fingertips, the operator can make sure that the cart machine is indeed executing its proper programming role.

LIBRARY MANAGEMENT

Library management in an automation environment is indeed a key factor in any cart machine with a high numerical cassette population. The current tendency to shorter and shorter commercial spot messages further emphasizes the need for precise, yet flexible library management. Of course good library management eases the task of handling the large number of cassettes that can be contained in the physical library. Three devices (heuristics) are employed to accomplish this goal. Exception reporting and run-time warnings inform the station staff of missing library events in the play list. Automatic Purge attempts to make room for required new material. Finally, Purge Protect attempts to prevent the automatic purge of material required in deference to new material being loaded.

As mentioned earlier, exception reporting consists of a printed listing of material both required and physically not present, as well as material required and unknown. It is for the purpose of a complete as possible exception report that there is a requirement from the automation system of a one-time transfer of the entire day's play list early in the broadcast day. It is anticipated that once this list has been generated, the automation system will send a shorter list that will actually be used to begin operation. If the exception reports are not desired, it is unnecessary to send the entire play list.

Run time warnings indicate that an event is missing and its time to air is less than a presetable time. The Immediate Load feature of the TCS2000 will maintain a clear path for this box number up until such time as it is too late to cue. By this, it is meant that both a VTR and a manipulator arm will be reserved. If the box is inserted in the load port and the Immediate Load button depressed, the machine will take as its highest priority job, that of fetching the box, loading and cueing the spot, in preparation for airing in its intended sequence. If the material is not loaded in time, the machine will either skip that event or, more logically for the automation environment, air Black and Silence for the nominal duration of the missing event.

Automatic Purge ensures that there are several free library pocket locations available. This is a requirement for immediate load; in order to accept an immediate load event under all conditions, a home pocket must be provided to send it to when it is completed. In addition, automatic purge allows Load to have a higher priority than Purge, so that an operator performing a load operation of needed material is not annoyed by waiting for a purge to occur first.

Purge Protect is required to ensure that automatic purge works smoothly. For up to a presetable maximum time, purge protect marks as "in use" with an "in use" count those elements in the play list residing in the library. These elements, along with items marked as "Permanently Purge Protected," are not subject to automatic purge. A library purge protect status is a function of the loaded play list, therefore, the purge protect status of individual elements is changed when a play list is modified.

GENERAL CART MACHINE PARAMETERS

A cart machine in an automated operational environment needs to meet a few basic requirements in order to maximize its utility to the station. One of the key factors is the number of separate cassettes that will be used to carry the commercial messages, or the program sequences for insertion into the normal program log.

The actual cassette population in a cart machine must be optimized for adequate daily programming capacity, without the burden of serious overkill that would make the machine physically unwieldy, and technically more complex. The ideal number of resident cassettes in a cart machine seems to be between 250 and 300, and the choice of 280 for the TCS2000 proved optimum for the robotic tower arrangement with its cart bay enclosure. With this number of cassettes in the machine, most stations can easily program a full eight-hour shift. The computerized data base, managing up to 65,000 spots and many, many 1,000 event play lists, takes care of most commercial station needs.

VTR formats are also an important factor in the design of a cart machine, since there are now at least three different cassette formats in use for other studio applications. Concurrently, the most widespread formats for cart machines are the Beta type L, and the half-inch VHS type M. However, type M has been displaced by the M-II format, which while being similar to M, is in fact incompatible enough to require a different VCR to handle it. The latest additions in the format field are the Beta SP, the digital D1 component, and the digital D1 composite. It is also conceivable that 8mm video cassettes, modified for broadcast operations, could represent a future cart machine potential. The cart machine described in this paper already has been operationally configured for the Beta, M and M-II formats, and can be reconfigured for any new formats that become practical.

SUMMARY

In general a well integrated cart machine, that is fully interfaced with the station's automation system, will provide the user with a smooth running and cost-effective means of handling daily program requirements. Fitted with the proper software modules, that optimize the interface between the cart machine and its information sources, it can provide the following functions:

1. A continuous job stream (play list) with "connectivity" between subsequent days, including a method of appending new material "on the fly."

2. A method of handling the cart machine's physical library conveniently as a subset of a much larger library (the station's library).

3. Input, editing, display and reporting (to a printer) of the library, the data base, and playlist data. Also a report of play list missing material and play list "aired" information.

4. Translation of Physical to Logical Material Identification number (house number). The bar code labels affixed to the boxes are professionally preprinted

and once affixed, remain on the box. The house number is assigned at the time of creation of the material (record) and can be preassigned to permit the inclusion of material not yet created into a play list.

The cart machine keeps a current "Library" file (like a data base) of all boxes that currently physically reside in the system. In addition it keeps a "Data Base" of all boxes ever known and not deleted, regardless of whether they are currently in residence in the system. Both the library and data base are "indexed" (to search for a particular event) on several "fields," including physical box number (box) and logical house number (house number or MID number). The play lists use information from both the library and data base. The data base provides such information as the title, default spot start time on tape (SOM), and default spot duration. The library provides the current physical location of the box.

In most station operations the cart machine capacity of 280 cassettes is sufficient to provide a full eight-hour operational shift without reloading. Stations with heavier duty cycles can of course set up their own sequence cycles on a shorter turnaround basis to provide continuous daily operations.

A P P E N D I X

The following are short examples of print outs covering various cart machine operations. A variety of other detailed printouts are also available on demand.

EXAMPLE OF LIBRARY REPORT

House #	Title	Box #	Loc	SOM	Dur	Plys	Last	PS
00000000		04955	G 10	00:00:00:00	00:00	000	00/00/00	N
03215	JELLO	03215	D 27	00:00:15:00	00:29	056	10/29/86	A
100090	TOYOTA	00159	G 35	00:00:15:00	00:30	001	06/11/86	A
100142	BOLENS	00372	F 36	00:00:15:00	00:29	009	02/03/87	A
100144	PIZZA HUT	00435	A 34	00:00:15:00	00:30	007	02/03/87	A
100176	SCHUELE PAINT	00164	B 33	00:00:14:03	00:30	005	02/03/87	A
100178	KELLOGS F.F.	00499	A 33	00:00:15:00	00:30	011	02/03/87	A
100212	TRUE VALUE	00348	B 34	00:00:15:00	00:30	004	02/03/87	A
100235	HILLS DEPT.	00429	F 37	00:00:15:00	00:30	001	06/09/86	A
100271	ALLSTATE AUTO	00417	B 36	00:00:15:00	00:30	001	06/11/86	A
100276	SODASTREAM	00181	G 30	00:00:14:00	00:30	016	07/09/86	A
100320	CHRYSLER	00338	A 32	00:00:14:10	00:30	003	06/04/86	A
100369	BLUE CROSS	00339	B 20	00:00:14:15	00:30	005	06/04/86	A

EXAMPLE OF DATABASE REPORT

House #	Title	ID #	Box #	Home LOC	Duration
100001	PAUL BATT BUICK	328	373	NONE	00:00:10:12
100002	RECORD TEST	2	4867	NONE	00:00:10:00
100002	RECORD TEST	3	4868	NONE	00:00:10:00
100003	CANNON	1225-1	124	NONE	00:00:29:15
100004	MARVIN F	33-1	282	NONE	00:00:30:00
100005	FAYS	116-1	126	NONE	00:00:30:00
100006	CLOISTER	58	127	NONE	00:00:09:15
100007	FAYS	146-2	121	NONE	00:00:30:00
100009	WISK	330	91	NONE	00:00:30:00

EXAMPLE OF EXCEPTION REPORT

Air Time	House #	Title	Box #	ID #	Status	E #
12:05:30	2292	FRISKIES	2292	JS2292	Non-library	1
12:28:03	2019	PETER PAN	7519	M REEL 1	Non-library	15
13:58:58	2058	SHEARSON LEHMAN	2058	CD1	Non-library	28
15:28:30	2151	RASIN NUT BRAN	2151	JM55	Non-library	40
15:45:00	2291	TALKING WRINKLES	2291	JJ2291	Non-library	47
16:01:59	2166	TOYOTA CAMRY WAG	2166	CD1	Non-library	62
16:31:15	2291	TALKING WRINKLES	2291	JJ2291	Non-library	70

EXAMPLE OF "AS RUN" LOG

Air Date	Air Time	House	Title	On Air	Remaining
02-10-87	10:23:26	2085	JORDACHE EARS	00:30:01	
02-10-87	10:23:56	2072	11 NEWS INTRO	00:31:05	
02-10-87	10:24:29	2137	NESCAFE SILKA	00:30:05	

AN INTEGRATED FACILITY FOR NEWS GATHERING AND BROADCAST — THE CBS HARD NEWS CENTER

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INTRODUCTION:

The volume of CBS News broadcasts and the complexity of gathering news, its editing and distribution, have grown substantially over the last several years. Today, thirty-three percent of all CBS Network programming consists of news and public affairs, or about forty hours per week.

The previously existing facilities for handling this volume of work were completely inadequate and physically inefficient, with much foot traffic and communication problems between the separated areas of activity.

The planning of a new facility was begun in 1980. Through close co-operation between CBS Operations and Engineering and CBS News' own operations department, the Hard News Center evolved in complexity and scope to both take advantage of new technologies and meet increasingly competitive demands. This paper outlines the requirements, planning, and real world operational experience in the new facility.

REQUIREMENTS:

While the Hard News Center was designed to service the needs of several different CBS News broadcasts, each broadcast had parallel goals:

(i) To support its broadcast in as efficient and contiguous a space as possible.

(ii) To adopt a universal tape format to simplify late feeding and editing.

(iii) To create a facility that would support the editing of a production intensive Charles Kuralt "On the Road" piece off-line, while recording, editing, and airing that night's Evening News broadcast (to take a specific example).

In engineering terms, the goals were translated as:

(i) To be able to receive and record up to 25 simultaneous news feeds from remote locations.

(ii) Provide both on-line and off-line edit capability. Provide the ability for off-line as well as on-line edit rooms to feed a control room.

(iii) Provide a special facility to rapidly edit and reformat the order of a particular broadcast up till and during air time.

(iv) Provide a central work station to monitor the rapid turn-around editing facilities as well as routing the work done in an off-line editing room to air.

(v) Integrate the entire production process with the newsroom editorial computer so that up-to-the-minute changes in broadcast order can be flashed not only to people but to machines.

In addition, a separate but contiguous control room was engineered to allow for integration of graphics, computer animations, and other production effects. Thus this control room (CR 34) acts also as a program building room.

THE NEWS FLOW PLAN

Figure 1 depicts the principal patns followed by a news feed enroute to broadcast. Up to twenty-five remote feeds may be received simultaneously at the Broadcast Center, and the incoming sources switcher routes the signal to one of twenty-five edit rooms where recording of the feed is performed.

Any feed can be observed in the Executive Producer Station where a decision on its inclusion in the broadcast is made. The feed may also be monitored in the International Communications Room which maintains communication with the remote reporter.

For a major story comprising a number of segments, the individual segments are passed to the Control Room 34 for integration together with graphics, stills, and legends.

The completed story on 1/2" Betacam cassettes is then taken to the adjoining Airplay/Record room (APR), ready for playback.

As shown by the serrated lines, a story that is complete and requires no program building, is taken directly from the edit room to the APR room.

The thinner flow lines indicate some optional routes. For example, a story that arrives completely edited in the field may pass directly from the incoming switcher to the control room for real time broadcast. Alternatively, it may be routed to the APR room and recorded directly ready for playback.

Some edit rooms and the Control Room 34 may also feed stories directly to the main Control Room 47.

Finally, the director in the control room initiates the playback of all stories and integrates them with the live anchor in Newsroom 47 prior to distribution to the network.

LAYOUT OF THE NEWS FACILITY

The new facility occupies the first three floors of the newest building at Broadcast Center.

On the lowest level is the Hard News Center; its basic layout is detailed in Figure 2. Twenty individual edit rooms, manned by CBS News editors, ring the late record/playback core. Three narration rooms, capable of being selected in any of the edit rooms, plus office space complete the perimeter. Seven of the CBS News edit rooms may be routed directly to air through the APR switcher.

The central facility contains multicassette machines for air playback, a bank of tape machines dedicated to the program building operations in CR 34, as well as two pairs of machines for recording of the first feed of a new broadcast and its subsequent playback as a second feed to the network.

Adjacent to the central area is the late record/playback facility and its accompanying five Betacam format edit bays. Manned by editors from CBS Operations, each bay contains two BVW-40 record/play machines and a switchable BVW-15 machine with freeze frame and slow motion capability. These bays known as Late Feed Edit Pairs, are designed to edit stories that are fed, via satellite, especially shortly before broadcast.

Adjoining the Late Feed Edit Pairs is the Associate Directors' work station, the central position from which the preparation and editing of all stories are monitored.

The International Control Room (ICR), with its facilities for coordinating and monitoring international satellite bookings, the CBS News maintenance shop, and the Hard News Center's switching racks round out the first floor.

The floor above the Hard News Center, occupying 9,000 square feet, contains the newsroom as shown in figure 3, which also acts as the news studio. The Evening News anchor's podium is seen in the center, surrounded by the national and foreign desks, wire service and secretarial desks.

The Executive Producers Stations (EPS) for the Evening News and Morning News, offices for the executive producer, the news anchor, and national and foreign editors, complete the amphitheatrical arrangement. The newsroom also contains a second set for the Morning News program while at the top right the control room is located adjoining a small flash studio.

The central area of the newsroom is open through the third floor, while around the periphery of this floor are offices for the producers and correspondents.

OPERATING EQUIPMENT

Each edit room is equipped with two half-inch Betacam VCRs and with a 3/4" Umatic VCR to accommodate those stories which are still occasionally delivered by hand in that format. The editing console used is the half-inch Betacam VCR built-in editor. The fact that an editor can view an incoming raw feed, enables him to gain an immediate overview of the material, and to start the editing task immediately.

When a story has been edited, the in-cue and out-cue points are located, and these points are entered directly into the barcode writer. The time code corresponding to these points is stored by the writer, and the duration of the story is then calculated and displayed automatically (Fig. 4).

The editor now has only to key in a given three digit number and a title which uniquely identifies the story. In addition, the editor also enters at the keyboard, the name of the news program, the date, the edit room, and the editor's initials. This information is not bar coded, but appears in alpha-numerics above the bar code. The cassette label is then printed and automatically affixed to the cassete, which is then ready for dispatch to the multicassette playback machine (Fig. 5).

For a story which is to be integrated with other segments, the cassette is taken to Control Room 34 where program building and integration takes place. The CR 34 is in fact a fully equipped control room and can function as such when, for example, the main Newsroom 47 control is engaged. In particular it can act as a control room for the flash studio for emergency bulletins.

Generally, news feeds which arrive close to air time are recorded on five additional Late Feed Edit Pairs (LFEP). The Late Feed Edit Pairs are equipped with three half-inch Betacam VCRs, (two BVW-40 VCRs and one BVW-15 for slow motion and still frame playback), and are controlled by a BVE-800 timecode editor. When edited, if time does not permit their insertion in the multicassette playback machine, the stories may be played to air directly under the command of the main Control Room 47 (Fig. 6).

An important feature of the Hard News Center is the Associate Director's work station. The work station consists of monitoring and communications to seven assignable edit rooms, the five Late Feed Edit Pairs, the three Betacarts, and both control rooms.

Thus the Associate Director can monitor progress in the preparation of all stories whether in the edit rooms, the Late Feed Edit Pairs or the program building room, and has voice communication with each.

Three Betacart multicassette playback machines are installed in the APR. The Betacart machine control system requires a barcode label on each cassette for operation. Cassettes are inserted randomly in the Betacart magazine as they become available. Upon insertion, the barcode label shown in figure 5 is read automatically and the data stored and displayed on the Betacart's data display unit. These units are located at each Betacart and in the control room. The director of the news broadcast can then see at a glance the identity and availability of all news stories, together with the order in which they are to play. The executive producer determines the sequence at which stories are to be broadcast and the play list for line up is entered on a news room computer terminal located in the Executive Producers Station.

The newsroom computer links the preparation process with the editorial. Beginning with the day's outlook, a broadcast is edited, changed, and updated constantly. Each one of these changes is flashed instantly via the Living Lineup to key terminals located throughout the Hard News Center. During the broadcast, the Living Lineup backtimes pieces and provides visual confirmation of on-air changes. Ultimately, this real time network of computers will direct not only the Betacart but an electronic prompter as well.

As tested, the Newstar computer system becomes the "brains" of the Betacart-replacing the operator's keyboard as the means of organizing the re-sorting playback sequences.

OPERATING EXPERIENCE

The new news facility has been in full operation since September 1986, and the following results have been achieved.

- The time taken to receive, edit and prepare a news story for broadcast has been reduced.

- Multi-segment news stories can be prepared more quickly, including the insertion of computer graphics, stills, and legends.

- The flexibility of the system and its ability to change a news lineup rapidly and efficiently, and to update a particular news story between the first and second network news feeds has been demonstrated.

- Foot traffic has been reduced and communications within the news facility greatly improved. In particular, the work station at which the status and progress of all stories in preparation can be monitored and directed has been a boon.

- Overall the productivity of the entire operation has increased, while material cost savings have been achieved by the use of half-inch cassettes in place of 3/4-inch Umatic and one inch reel-to-reel machines.

- The new facility can accommodate all the current normal and crisis needs of CBS News, and can be readily adapted to meet future news editing and distribution service requirements.

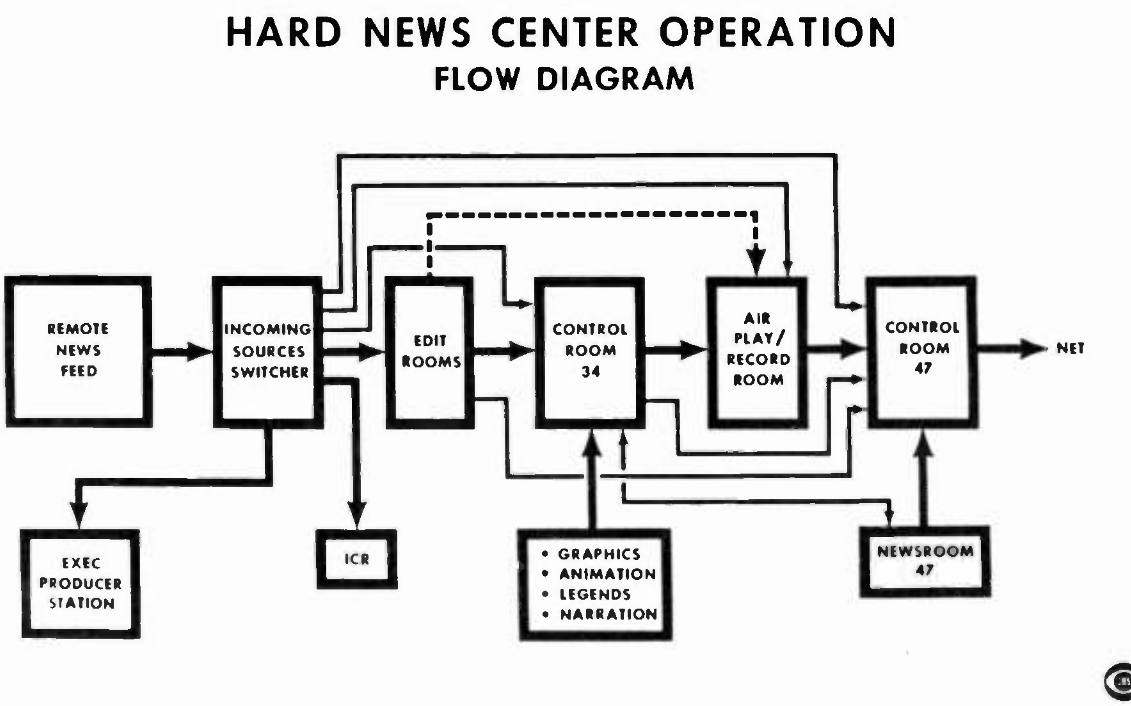


FIGURE 1.

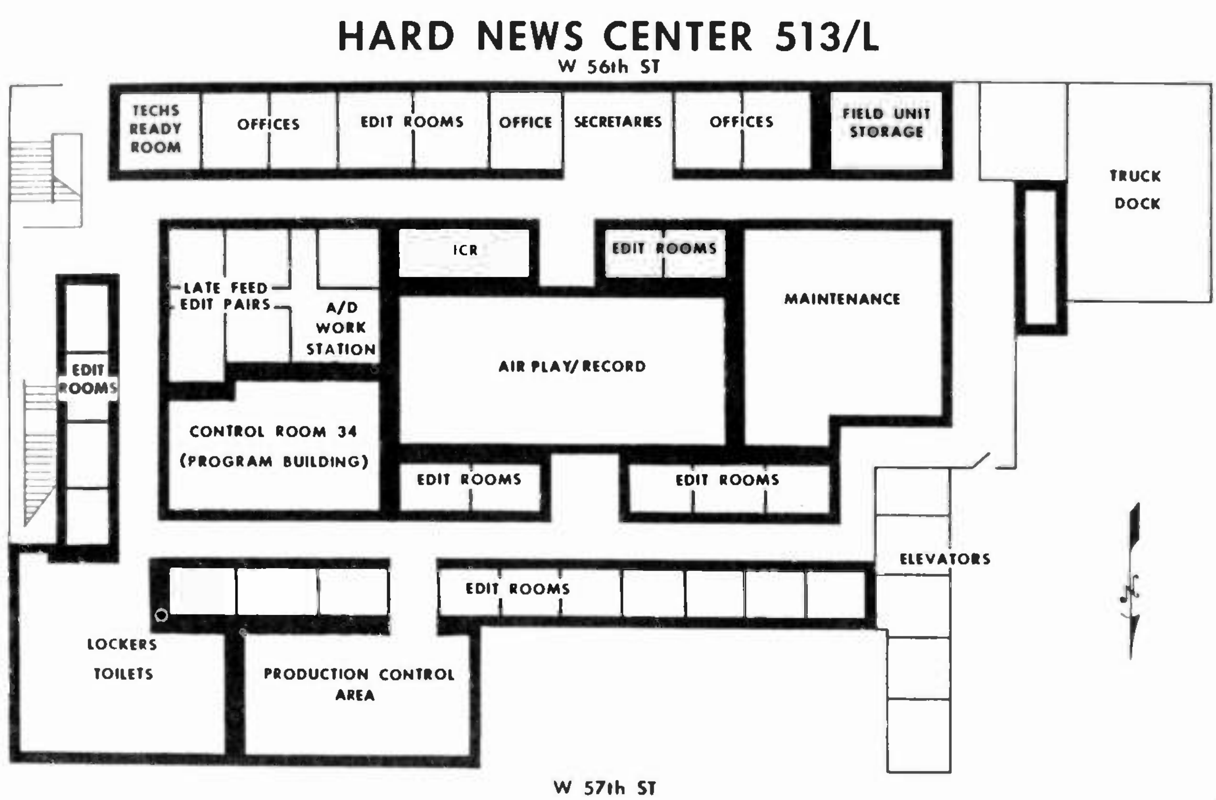


FIGURE 2.

NEWSROOM 47 513/1

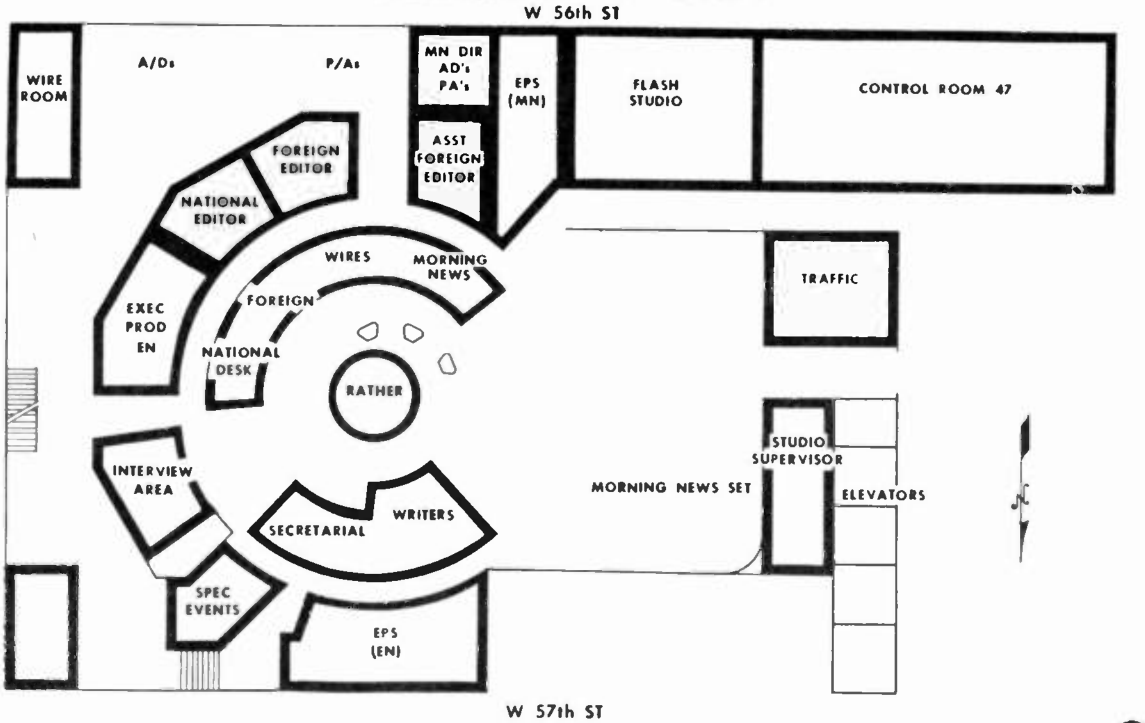
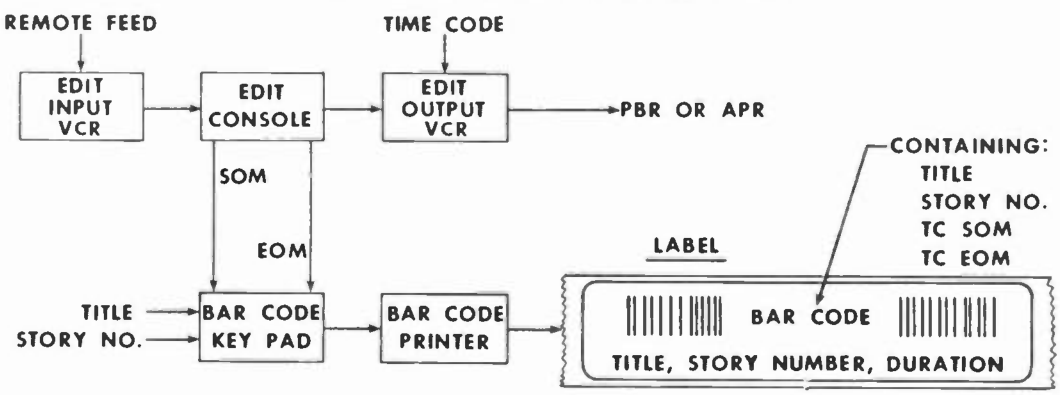


FIGURE 3.

EDIT ROOM OPERATIONS



OPERATOR ACTIVITIES:

1. EDIT STORY
2. PLAY BACK STORY
3. WITH SINGLE KEYSTROKE, RECORD SOM & EOM
4. ENTER STORY TITLE & NUMBER ON KEY PAD
5. ENTER "PRINT" ON KEY PAD

SYSTEM FUNCTIONS:

1. EDIT CONSOLE AUTOMATICALLY TRANSMITS TC SOM & TC EOM TO BAR CODE UNIT
2. BAR CODE UNIT CALCULATES DURATION FROM SOM & EOM DATA
3. ON COMMAND "PRINT", BC PRINTER PRODUCES A LABEL CONTAINING
   TITLE, NUMBER, & DURATION AND BAR CODE CONTAINING
   TITLE, STORY NUMBER, TC SOM & EOM

FIGURE 4.

BAR CODE LABEL

SHOW'S NAME - DATE - VT OR ED RM - EDITORS' INIT'S S
ID# TITLE SOM-01:00:00:00 DUR-00:00:32:00

SHOW'S NAME - DATE - VT OR ED RM - EDITORS' INIT'S S
ID# TITLE SOM-01:00:00:00 DUR-00:00:32:00

FIGURE 5.

AIR PLAY/RECORD ROOM

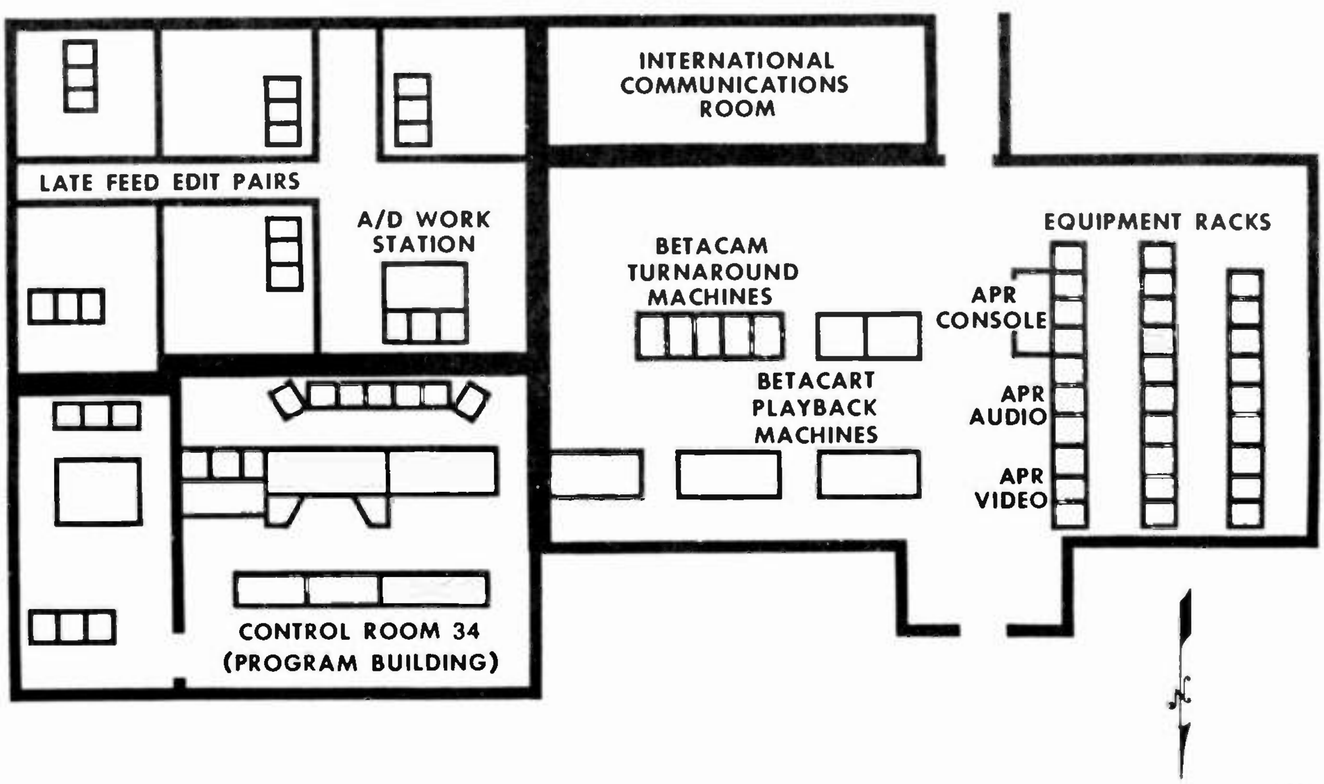


FIGURE 6.

TRENDS IN TELEVISION AUDIO—NOW AND IN THE FUTURE

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Abstract

The electronic sophistication of the American public, the competitive nature of the video marketplace, and recent technological innovations have generated and sustained a keen interest in high-quality stereophonic television audio. We have come a long way in a few short years and ongoing developments are creating an exciting future for the sound portion of the aural and visual medium of television.

For much of United States television broadcasting's history, the sound that accompanied the visual image was accorded the status of "second class citizen" and, indeed, sometimes appeared to be little more than an afterthought. Our system of transmitting television aural signals employs frequency modulation and has thus always contained the potential for high-fidelity sound transmission, and there was ample bandwidth available for the inclusion of subcarriers to support multichannel sound. That television audio failed to realize its potential for so long may be attributed to a number of causes. Terrestrial network delivery systems compromised the quality of audio delivered to affiliates. Video tape recorders were optimized for video performance to the neglect, if not the outright detriment, of their audio performance. Studios and sound stages were often acoustically deficient and noisy. Television receivers had "low-fi" sound sections consisting of anemic power amplifiers driving small, low-quality speakers. These receivers delivered unexciting sound to the viewer, but did offer the benefit of filtering out the buzz created by such transmission impairments as incidental carrier phase modulation and visual carrier overmodulation.

These factors, coupled with a general lack of interest by the industry in fully exploiting the potential of television audio, caused the sound portion of the aural/visual medium of television to languish for several decades. Through the years there had been some inquiries regarding stereo sound for television, beginning in 1959-60 along with an FM stereo broadcast study, but each inquiry was met with a tidal wave of indifference by the television industry and quietly faded away.

In recent years, the perspectives of both producers and consumers of United States television hardware and programming have been changed by a number of developments. The standardization and nearly universal employment of video tape formats containing multiple audio tracks has been coupled with the improvement of these tracks to reasonable, if not exemplary, quality. Satellite network and syndicated program distribution facilitates the inclusion of multiple high-quality audio channels with the visual signal. The proliferation of high-fidelity stereo consumer audio equipment has spilled over into the video sector, and pre-recorded videocassettes with theatrical-quality stereo soundtracks are as close as the corner video store. Competition among the video media from such sources as independent broadcasters, cable television, videocassettes and videodiscs, and direct satellite delivery has multiplied. Broadcasters are searching for ways to more fully exploit their television signals, and the color receiver market has matured. These factors have acted in concert to fuel the industry's interest in high-quality multichannel television sound. This brought about re-examination of the MTS question in the early 1980s and led to the development and implementation of the BTSC system for transmitting multichannel television sound, a watershed development in broadcast television audio in the United States.

The rapid acceptance and widespread awareness of broadcast stereo television and the availability of high-quality stereo sound on other video media has spawned efforts on the part of broadcasters and receiver manufacturers to improve the fidelity of television audio from production through the broadcast chain all the way to the living room.

To the public, stereo and high fidelity are so inseparable that they are virtually synonomous. The viewer who is enjoying television in stereo is either using a television receiver with a reasonably high-quality audio section or is routing the stereo television audio through the home high-fidelity system and, in either case, will be only too aware of any fidelity problems in the transmitted audio signal.

The result has been a ripple effect in television audio equipment design and practices which is producing improvements in audio distribution and processing equipment, in aural transmitters, and in television audio systems engineering. New attention is being paid to such factors as distortion, signal-to-noise ratio, and audio bandwidth in both transmitting and receiving equipment.

The audio side of television production is benefitting from the current interest in improving television sound as well. Studio acoustics are receiving new attention. Audio control room acoustics and speaker placement are being scrutinized, and audio consoles featuring impressive performance specifications and highly automated functions are being installed.

A renewed interest in some venerable stereo microphone techniques has surfaced. New versions of the X-Y and M-S type of far-field microphones are appearing which are readily adaptable to television boom miking and field pick-up situations and because they use co-located elements, their stereo signals inherently sum to mono without phase cancellations.

The video tape recorder has been identified as a weak link in the television audio chain, and work is proceeding apace to improve its signal-to-noise ratio, headroom, and distortion characteristics. A prime enemy of stereo television broadcasting is inter-channel audio timebase or phase error, which can result from azimuth misalignment of longitudinal audio head stacks. Changes are being made to current types of video tape recorders to enable routine azimuth adjustment, and some devices have been developed which electronically correct these time delay errors between stereo tape tracks as well. The efforts to correct or eliminate audio timebase errors are critical because such errors degrade the mono sum signal, something no broadcaster can live with.

Along with new video tape formats, some new techniques of recording audio on videotape have recently appeared. These include such developments as noise reduction for longitudinal audio tracks and the incorporation of companded FM audio channels, both of which are available on the new half-inch professional video formats. These FM channels provide wide dynamic range and low distortion, with the disadvantage that they cannot be edited separately from the video.

A development holding great promise for the future of audio on videotape is the employment of PCM digital audio recording for both one-inch and the new MII formats. Sixteen-bit linear quantization and 48kHz sampling provide complete audio transparency, total editing flexibility, multigenerational audio dubbing without degradation, and the absence of time delay or phase errors between stereo channels.

The word in television audio's future is increasingly going to be "digital". The digital trend, already underway, will develop into a major tidal wave, washing over all facets of television audio from production to broadcasting to reception.

In addition to magnetic tape, digital audio will be stored in such media as magnetic disc, solid state memory, and optical disc. In production and post-production, dramatically increasing usage will be made of non-tape based digital audio editing systems, which run the gamut from disc-based emulations of audio cartridge machines to such wonders as the audio sampling synthesizers, which are editing systems and musical instruments integrated into one device. These synthesizers can store and manipulate sound in a dizzying variety of ways, in synchronism with video signals.

Digital audio processing is proliferating in the form of delay and reverberation devices, pitch changers, and time compression devices, all of which are characterized by impeccable audio quality. The future will bring us fully digital audio consoles in which the functions and parameters of audio processing modules will be software-controlled. What the modules do functionally will be limited not by their nature as hardware, but by computer processing power alone; and as an additional bonus, these consoles will diagnose their own faults and reroute signals around malfunctioning sections! As we move into the future, an ever greater amount of the audio editor's work will be done using a computer terminal.

Distribution of audio, both within the plant and to the outside world, will increasingly be done in digital form. The digital domain has some very attractive and useful qualities for television audio, including wide dynamic range, very low distortion, robustness under adverse conditions, and the capability to withstand many generations of recording and many stages of processing and distribution without significant degradation.

The day will come when television audio will remain in the digital domain virtually from its origin to the receiver. Digital audio delivery to the home will give the consumer the ultimate in audio quality from the television set.

The near future will increasingly bring the augmentation of stereo with surround-sound, a natural adjunct to the video presentation, particularly with the expanding presence of large-picture and projection televisions, and most especially to accompany high-definition and wide-screen formats. This will be the next audio step toward bringing the theater experience into the living room. A future generation of television receivers will, no doubt, include integral surround-sound decoders.

In conclusion, the increasing electronic sophistication of the audience, as well as the producers and disseminators of television programming, in combination with the maturation of the television medium and increasingly competitive nature of the video marketplace, have generated and sustained a keen interest in high-fidelity and stereophonic television audio. The cornucopia of new developments in audio for video which are being used and developed by the television industry is creating an exciting future for television audio. At this point, we have only seen the beginning!

STEREO AUDIO TRANSMISSION ON SATELLITE AND TERRESTRIAL LINKS

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Stereo Television is gaining worldwide momentum. Slowly but steadily the number of consumers being exposed to enhanced television audio is growing. While broadcast transmission standards vary, the common problem of transmitting high quality stereo audio over satellite and terrestrial wideband facilities must be solved. This paper will explore alternatives available to system planners as well as discuss several standards in use today.

There are three generalized elements to stereo TV: 1) Production, 2) Distribution, and 3) Transmission to Consumers. Element 1, production, is not within the scope of this paper. Element 3, transmission to the consumer, is defined by various standards throughout the world. The main consideration in delivery to consumers is compatibility with the massive universe of existing monaural TV sets. The problem of stereo distribution is totally different than that of transmission to consumers.

Stereo distribution via satellite and terrestrial microwave is most commonly done with subcarriers - narrowband signals that occupy the upper part of a baseband signal above video. While subcarrier audio channels in themselves are nothing new, their use for high quality stereo with today's excellent source material requires careful analysis.

Five Different Systems

As of this writing, there are at least five different techniques being used to transmit stereo audio via satellite and microwave. They are:

1. Discrete Left and Right Channel High Level Subcarrier

2. Discrete Left and Right Channel Low Level Companded Subcarrier

3. Dual Subcarrier Matrix System Using Two High Level Subcarriers

4. Multiplexed Matrix Single High Level Subcarrier

5. Spectrum Efficient Digital Subcarrier

Discrete Left and Right Channel High Level Subcarrier transmission is perhaps the simplest and most straight forward technique. In this system, two subcarriers are used as a stereo pair. Typical parameters include a deviation of 237 kHz peak at PPL (75 kHz at OVU) and a subcarrier injection level of 100 mv peak to peak prior to pre-emphasis by the video modulator. For mono compatibility, the receive location must simply sum the signals together if stereo is not being used. Reasonably good quality audio can be delivered with this system with signal to noise ratios in the order of 70 dB (relative to Peak Program Level) being the norm. Since the Left and Right Channels are totally independent, the stereo separation is very high - typically 60 dB or greater. The main disadvantage of this system is the relative inefficiency with respect to maximum loading of the baseband. In some cases this is important, in others it is not. The NBC Television Network is a good example of high quality stereo audio being delivered to affiliates with discrete Left and Right channel subcarrier technology.

Discrete Left and Right Low Level Companded Subcarrier technology has been a widely accepted means of transmitting stereo audio since 1980. Virtually all U.S. cable programmers are using this technology as well as several TV syndicators delivering programming to affiliates. The main advantage of this technology is the delivery of equivalent or better audio performance with a minimum amount of subcarrier spectrum and power. Typically, the peak deviation is 50 kHz at PPL with a subcarrier injection level 4 to 6 dB lower than high level carriers. This allows more efficient baseband loading, and less impact on video performance. Recovered audio quality is every bit as good and in some cases better than high level subcarriers. Signal to noise ratios of between 70 and 90 dB (relative to Peak Program Level) are the norm and stereo separation is always greater than 60 dB. The only real disadvantage of this technology is the increased circut complexity necessary to recover the processed audio. In the broadcast world, examples of people using this technology are Robert Wold Company and TVSC.

Dual Subcarrier Matrixed systems rely on two high level subcarriers; one for L+R and one for L-R. This system gained widespread acceptance on satellite networks with a large installed base of mono receive locations where stereo would be an add-on option. In this case, L+R audio is transmitted on 6.8 MHz and L-R is transmitted at 5.8 MHz. With the signals transmitted in a matrix format, gain and phase matching becomes critical. Dual demodulators and a matrix must be carefully adjusted to obtain reasonable separation.

Performance of this type of system can also be very good with typical signal to noise ratios of approximately 70 dB (relative to Peak Program Level) and stereo separation greater than 30 dB. No broadcast programmer is presently transmitting with this format. Several cable TV programmers do however, including MTV and The Disney Channel.

The multiplexing of a stereo signal onto a single subcarrier has proven to be a technology of limited value for high quality transmission. Just as it does in FM stereo multiplexing, the difference between mono and stereo signal-to-noise ratios is very substantial due to the L-R signal being placed high in the baseband. To the best of the author's knowledge, this technology has been totally abandoned for satellite stereo transmission. It has recently been suggested that a variation of this technology, BTSC with enhanced deviation, might be useful on satellite circuits, but transponder loading and signal to noise ratio performance will likely make it unpopular. Later in this paper, we will see why this is so.

Spectrum Efficient Digital Subcarrier transmission systems are starting to emerge. Digital systems such as 16-bit PCM are just too bandwidth inefficient to be placed directly on a subcarrier. Several techniques for reducing the required bandwidth are currently available, the most popular being that developed by Dolby Laboratories. This clever system can transmit relatively high quality stereo audio with as little as 512 kbits/second. Such digital transmission systems are capable of 80 dB or greater signal to noise ratios (relative to Peak Program Level) with injection levels in the same range as high level analog subcarriers. A good example of programmers using Dolby Digital Transmission are U.S. cable networks MTV and VH-1.

Subcarrier Performance Analysis

In the preceding general discussion about the various transmission systems, you will notice that all had a comment or two regarding signal to noise ratios. This is generally the one most path-dependent parameter with such items as frequency response, distortion, etc. being determined by actual hardware elements. I have found it very useful to be able to quickly analyze link performance with respect to signal to noise. If you can determine the basic received C/N for a satellite downlink or microwave hop, the subcarrier performance can be quickly and logically analyzed.

The first thing one must do is derive the C/No received by the main demodulator. This is obtained from:

$$C/No = C/N + 10\log BW \qquad (1)$$

For example, if you have a satellite downlink receiving a C/N of 12 dB and the receive IF bandwidth is 30 MHz, the C/No = $12 + 10\log(30\text{x}10^6) = 86.8$ dB-Hz.

The next thing to determine is the subcarrier C/No which is defined by:

$$(C/No)_{SC} = C/No + 10\log(m^2/2) \qquad (2)$$

Where m = modulation index of subcarrier

For example, if you have a 6.8 MHz subcarrier and it is modulating the main carrier at 2.0 MHz, the modulation index = 2.0/6.8 = 0.294. Thus :

$$(C/No)_{sc} = 86.8 + 10\log(.294^2/2) =$$

$$73.2 \text{ dB-Hz}$$

By the way, a typical subcarrier injected at 100 mV P-P will actually be boosted 3.5 dB by the pre-emphasis circuit in the NTSC video modulator to approximately 150 mV P-P. The actual modulation index can be determined by the ratio of maximum video voltage (1V P-P) to the main video deviation.

For example, if a 1.0V P-P signal provides a peak deviation of 10.75 MHz, then the subcarrier deviation will be:

$$\frac{1.0}{.15} = \frac{10.75}{X}$$

$X = 1.6125$ MHz Peak Deviation

Now, we can determine the audio signal to noise ratio. Assume we are looking at a high level subcarrier with a peak deviation of 237 kHz at Peak Program Level (75 kHz + 10 dB "Headroom"):

$$(S/N)_{audio} = (C/No)_{SC} + 10\log[3(237000)^2/2(15000)^3] + P \quad (3)$$

where P is the pre-emphasis advantage for FM, which for 75 μ sec. ≅ 12.3 dB. Thus:

$$(S/N)_{audio} = 73.2 + (-16) + 12.3 = 69.5 \text{ dB}$$

This says that the Peak Program Signal to unweighted noise level should measure 69.5 dB for this example. This calculation compares favorably with actual field data.

Notice that the controllable variables in this example are simply subcarrier injection level (m) and audio deviation.

One must always be sure that the subcarrier demodulator is always operating above threshold for any of the performance assumptions to be valid. A close approximation of the threshold point can be determined by the formula:

$$\text{Threshold } C/No = 10 + 10\log BW \quad (4)$$

where BW is the noise bandwidth of the subcarrier demodulator.

Going back to our example of a high level subcarrier, the noise bandwidth should be at least 500 kHz, thus the threshold C/No = 10 + 57 = 67 dB-Hz. This is approximately 6 dB above the predicted operating level, and all equations will be valid.

Low Level Subcarrier Performance

The preceeding discussion is valid for low level subcarriers, but the parameters are different in terms of peak deviation and modulation index. Typical peak deviation is 50 kHz and typical modulation index if 0.14. This obvious loss in performance is immediately made-up with audio companding. Thus equation 3 can be re-written for low level subcarriers to include a factor for compander performance. Thus:

$$(S/N)_{audio} = (C/No)_{sc} + 10\log[3(50000)^2/2(15000)^3] + 12.3 + Ac$$

For Wegener Panda\ I System, Ac = 20 dB
For Wegener Panda\ II System, Ac = 40 dB

If you go through the same examples of a video receive operating at a C/No of 12 dB, you will find the audio signal to noise ratio is:

69.5 dB for Wegener Panda\ I and
89.5 dB for Wegener Panda\ II.

These numbers correlate very closely with actual perforamcne. Narrowband low level subcarrier can comfortably compete in terms of audio quality and will win hands down in terms of baseband power loading and spectrum efficiency.

Multiplex Subcarrier Analysis

The performance of a composite multiplex subcarrier must be broken down into two parts: L+R and L-R.

L+R Signal-to-Noise can be defined as:

$$(S/N)_{L+R} = (C/No)_{sc} + 10\log[3(DEV)^2/2(Fm)^3] + P$$

where DEV = Peak Deviation at PPL
Fm = Highest Modulation Frequency (15000 Hz)
P = 75 μ sec Pre-Emphasis Advantage ≅ 12.3 dB

L-R Signal-to-Noise can be defined as:

$$(S/N)_{L-R} = (C/No)_{sc} + 10\log[(DEV)^2/4(Fmx)^2(fm)] + P$$

where DEV = Peak Deviation Caused by DSBSC Signal at Peak Program Level
Fmx = Center Frequency of DSBSC Signal (31468 Hz for BTSC)
Fm = Highest Modulation Frequency (15000 Hz)
P = Pre-Emphasis Advantage for DSBSC Signal (NOT FM) ≅ 6.5 dB

For an example, let's assume that a BTSC signal were to be transmitted on a 6.8 MHz subcarrier and the main carrier C/N = 12 dB. Going briefly through the total analysis:

$(C/No) = C/N + 10\log BW = 86.8$ dB-Hz

$(C/No)_{sc} = C/No + 10\log(m^2/2) =$

73.2 dB-Hz (for $m = 0.294$)

$(S/N)_{L+R} = (C/No)_{sc} +$

$10\log[3(25000)^2/2(15000)^3]+$

12.3

$= 73.2 + (-35.6) + 12.3 = 49.9$ dB

$(S/N)_{L-R} = (C/No)_{sc} +$

$10\log[(50000)^2/4(31468)^2$

$(15000)] + 6.5$

$= 73.2 + (-43.8) + 6.5 = 35.9$ dB

In addition, for BTSC, the L-R signal is companded with a companding advantage of approximately 14 dB, thus $(S/N)_{L-R} = 35.9 + 14 = 49.9$ dB.

Matrixing of the L+R and L-R signals each with an S/N = 49.9 dB yields a net S/N ratio for L and R channels of 46.9 dB.

The point of all this is to explain why BTSC directly on a satellite subcarrier will provide marginal performance at best. The predicted S/N ratio of 46.9 dB is relative to Peak Program Level - relative to OVU, it would be 10 dB lower (36.9 dB) - which is totally unacceptable performance in view of today's high quality source material.

Some have suggested increasing deviation approximately 6 dB as a solution to transmit BTSC directly. This will still provide marginal performance (S/N Left or Right = 43 dB) and require precise composite demodulation in order to maintain stereo separation.

Gain and Phase Error vs. Stereo Separation

Once a signal is matrixed to L+R and L-R, very accurate gain and phase control must be maintained to insure that stereo separation will be acceptable. Note that the gain and phase accuracy requirements only apply to the L+R and L-R signals, not individual Left and Right channels.

For example, discrete Left and Right channel subcarrier systems keep the two channels totally isolated and deliver Left and Right signals to the BTSC encoder at the transmitter with practially unlimited separation (typically 60 dB or greater).

Variations in subcarrier amplitude have no effect whatsoever on stereo separation. Small amounts of gain and phase error in the Left or Right channels will simply be tracked through the BTSC system and will not affect overall separation of BTSC signals. As an example, the CCIR has made recommendations for circuits designed to haul high quality stereo signals. Recommendation 505-2 allows gain errors of up to 3 dB and phase error up to 40 degrees at 15 kHz. Commercially available subcarrier systems are much better than these recommendations permit.

STL Links vs. Satellite Links

The preceding audio subcarrier performance analysis examples are valid for both types of links. The main difference in the links is in terms of normal operating C/N. Satellite links are quite predictable and relatively constant - especially C band. Fade margins are generally small compared to those on a typical STL link. Even a very conservative Ku band video downlink with 15 dB of fade margin is typically operating much closer to threshold than an STL microwave link at most times. When an STL link is operating well above threshold (30 to 40 dB), the noise performance of any subcarrier system will be more than adequate. As the path fades, the composite system will become dramatically worse at an earlier point than either high level or low level discrete subcarrier systems.

The main objective of the STL link is to convey the video and audio signals to the TV transmitter with little or no degredation. Up to the point of the final TV transmitter, all signals must be kept as clean as possible to insure that all significant degredations are caused by parameters that are outside the control of the broadcaster (RF path, cable systems, receive antennas, receiver demodulator, etc.). By transmitting all audio signals as separate components and locating the BTSC generator at the transmitter, this condition is satisfied.

To illustrate the point, if we compare three subcarrier systems on an STL link as a function of fade margin, it will be obvious that a composite BTSC signal could very well become the weak link in delivering high quality audio to viewers.

As the STL path fades, the composite subcarrier system will provide an increasing amount of L-R noise to the BTSC signal that the consumer's decoder must cope with. This is not the case with discrete subcarrier systems which maintain very high level audio S/N ratios at low receive levels.

Conclusion

The main controllable element in satellite and STL audio transmission is signal to noise ratio. The purpose of the paper has been to explain some of the logic behind the selection of discrete L and R subcarrier transmission as opposed to composite multiplex. The BTSC system is a compromise - a very clever compromise to provide compatible transmissions to consumer TV sets. Our objective should be to deliver virtually "straight wire" perforamnce up to the point that the compromise must be taken. Since this issue involves getting the signal to the TV transmitter, we are not limited to transmission standards such as FM stereo or BTSC and are free to optimize the transmission link method for transparency as opposed to compatibility.

A COMPATIBLE SURROUND SOUND SYSTEM FOR THE BTSC-MTS STEREO TELEVISION SYSTEM

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ABSTRACT

The BTSC-MTS (Broadcast Television Systems Committee-Multichannel Television Sound) Stereo Television system presently lacks a practical means of providing a discrete or "hard" center channel. Details of a 4-2-4 amplitude/phase matrix encode-decode system based on the Dolby Stereo two-channel motion picture audio process are described. The resulting system provides the addition of both discrete-like center and surround channel outputs and can be demonstrated to be compatible with mono and two-speaker stereo reproduction. Considering the fact that two high quality transmission and storage channels are rapidly becoming the standard audio complement to television and video, this process is a logical means of maximizing their effectiveness. Practical considerations regarding broadcasting and consumer decoding and reproduction are discussed.

INTRODUCTION

The BTSC-MTS Stereo Televison system is growing rapidly in terms of program production, consumer receiving equipment, and consumer awareness. With this growth comes both problems and opportunities. Many problems are based on technique rather than technology and are being effectively dealt with as system experience is gained.[1] The lack of a hard or discrete center channel is, however, a technological problem that has not been addressed by the MTS stereo system and is observed to be a distraction to viewing pleasure by many consumers. A solution to this problem is available today based on a professional system (Dolby Stereo) developed for optical motion picture two-channel soundtracks. This system makes use of a 4-2-4 amplitude-phase matrix employing steering or matrix enhancement techniques during the decoding process.[2] In addition to providing a discrete-like center channel output, this system also provides a surround or rear channel output allowing for a variety of left-to-right, front-to-back, and surround audio effects. Program material produced using this system is presently available to consumers on video disc and video tape software, direct satellite broadcast, and to a limited degree via MTS broadcast stereo television and cable television FM stereo simulcast (Figure 1). Programming in these formats is commonly referred to as being reproduced in Dolby Surround, which is a technical subset of Dolby Stereo.

The AUDIO VIDEO CHAIN

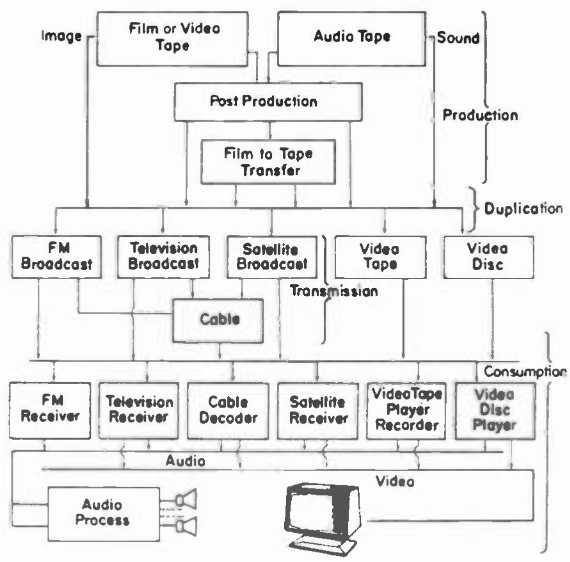

Figure 1

A number of consumer decoders are presently available to decode this program format with varying degrees of accuracy.[3] Such decoders are generally referred to as being steered or unsteered in their decoding action. This paper will explore the technology involved in this audio process with regard to stereo and mono compatability, MTS broadcast considerations, and consumer reproduction requirements. In particular, the details of a steered or active decoder processor will be described.

PROBLEMS AND LIMITATIONS WITH CONVENTIONAL STEREO TELEVISION

It is rather interesting to note that television viewing in the traditional sense has always had the sound source (speaker) in close proximity to the viewing screen, thus fusing so to speak, the audible and visual images. Viewers have, consequently, grown to expect, perhaps unconsciously, that this is the normal mode of watching television. On the other hand, the advent of stereo television using two discrete channels has significantly changed this perception as a direct consequence of the resulting loudspeaker placement. By positioning loudspeakers,1' or 2' to the left and right of the screen, centrally positioned images are only perceived as such for viewers positioned equal distance from the two loudspeakers. As one deviates from these locations, the sound appears to originate from the closest loudspeaker position. A common solution to this problem is to move the two loudspakers in towards the television screen which is a definite improvement albeit at the expense of significantly reducing the stereo sound stage.

In contrast, this has not been a problem with audio only systems due to the fact that most listening is more casual and lacks a visual frame of reference. Serious listeners, of course, tend to sit in the middle and take advantage of phantom center images.

An effective solution for television stereo would be to add a third centrally placed discrete loudspeaker channel. This would combine the best of both worlds but, unfortunately, is not possible with two-channel program storage and transmission systems.

A practical solution to this problem has been developed by the motion picture industry which faces the same type of problem in the theater due to the wide seating area. This process is a part of the system commonly known as Dolby Stereo and involves a matrix process designed to encode four audio channels onto a two-channel format. This process is shown in block diagram form in Figure 2, where it is noted that an additional surround channel is available to allow the creation of more complex sound fields. This process has been refined over the past ten years and used to produce over 1,000 major motion pictures involving a wide variety of music, dialog, and sound effects.

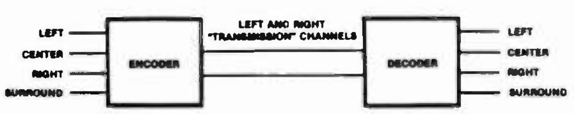

Figure 2

In order to gain insight as to why the motion picture matrix process has evolved the way it has, and as to why it is particularly well-suited for the consumer environment, it is instructive to look at the theater loudspeaker arrangement (Figure 3). Assuming that the four channels used (L,C,R, and S) are sufficiently discrete, this arrangement of loudspeakers provides a number of important production capabilities:

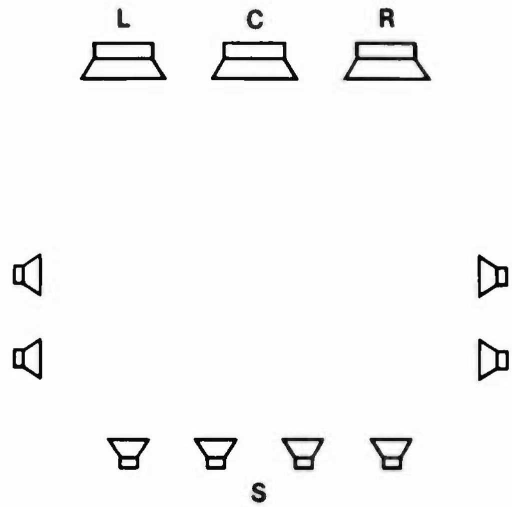

Figure 3

Capability 1

Discrete image localization for a wide listening area from three frontal loudspeaker positions.

Typical Use--Principal dialog, off-screen voices, music, and sound effects.

Capability 2

An interior or all around image produced simultaneously by all loudspeakers.

Typical Use--Environmental sounds such as wind, rain, interior building, car, or airplane sounds intended to create an atmosphere or mood.

Capability 3

Motional sounds resulting from panning across the frontal soundstage or between front and back locations.

Typical Use--Moving sounds such as cars, trains, airplanes, and space ships moving across the screen or between front and back locations.

Capability 4

Simultaneous left and right or front and back sounds.

Typical Use--Stereo music and sound effects, simultaneous dialog and background sounds.

Capability 5

Combination of interior and localized, panned, or simultaneous sounds.

Typical Use--Full action sounds involving combinations of music dialog and sound effects.

Collectively, these capabilities provide a very powerful means of enhancing the total audio visual experience.

THE MOTION PICTURE MATRIX PROCESS--TECHNICAL DESCRIPTION

The matrix portion of the Dolby Stereo process is a non-symmetrical encode/decode process in the sense that the decoding process involves adaptive matrix modification circuits to make the system sound as if four independent discrete channels are being used.

Figure 4 indicates the basic elements of the encoding process in block diagram form, which may be described mathematically as:

$$L_T = L + .707\,(C - jS) \qquad (1)$$
$$R_T = R + .707\,(C + jS)$$

where L_T and R_T represent the total left and right encoded signals, and L,C,R, and S represent the input left, center, right, and

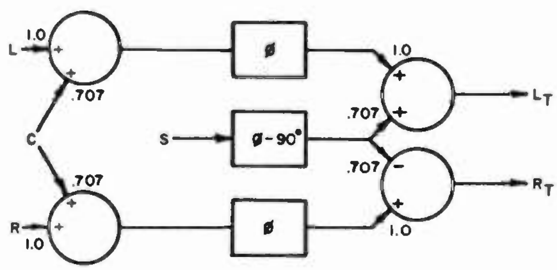

Figure 4

surround signals respectively. It is important to note that the elements in the block diagram denoted by ø and ø-90° are all pass networks characterized as having a flat amplitude response, but a frequency dependent phase response of ø (f) and ø-90°(f). Analysis of the complete set of encode/decode equations reveals that such phase shift functions are required in order to allow for an interior or all around sound capability.

The decoding matrix (before directional enhancement or matrix modification) is defined by the four equations:

$$L' = L_T \qquad (2)$$
$$R' = R_T$$
$$C' = .707\,(L_T + R_T)$$
$$S' = .707\,(L_T - R_T)$$

where the prime notation is used to denote that the decoded signals are only approximations to the original L,C,R, and S inputs.

The need for matrix enhancement or modification becomes clear when the decoded signals (2) are represented in terms of the original input signals. This can be seen below where equations (1) and (2) have been combined to produce:

$$L' = L + .707\,(C - jS) \qquad (3)$$
$$R' = R + .707\,(C + jS)$$
$$C' = C + .707\,(L + R)$$
$$S' = -jS + .707\,(L - R)$$

These equations clearly show that each decoded speaker signal contains the original input signal as well as its adjacent speaker's intended signals at 3 dB attenuated levels. This can also be shown graphically as in Figure 5 where the 3 dB arrows indicate the areas of significant crosstalk and the infinity arrows indicate channels producing no crosstalk.

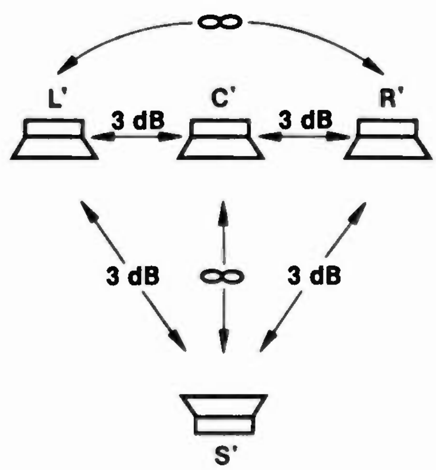

Figure 5

IMPROVING LOCALIZATION ACCURACY--DIRECTIONAL ENHANCEMENT

From a perception standpoint, the situation depicted in Figure 5 is a step in the right direction in that all reproduced sounds are most dominant in the intended directions. As a matter of fact, most consumer surround sound products available today provide this level of performance accuracy. Subjectively, however, such systems cannot produce unambiguous localization perception over a wide listening area, particularly with respect to centrally placed images.

A number of techniques have been developed to deal with this limitation and are commonly referred to as matrix enhancement, steering, or active circuits.[4,5,6] Functionally, such circuits serve to reduce the undesired crosstalk signal components as a function of the dominant content of the program. Since no technique has yet been developed that produces four independent channels matrixed into two, it is always necessary to make system compromises. In the case of the Dolby Stereo system, compromises have been made with artistic considerations in mind. For example, the integrity of the center channel is most important, and a number of steps are taken to keep center channel sounds, such as dialog out of the surround channel. On the other hand, the ability of the system to simultaneously reproduce a center and right or left discrete image are not very important and, consequently, are not possible without perceived crosstalk. The system can, however, very quickly alternate between these locations giving the impression of discrete behavior. As a result of such compromises, it is common practice to monitor Dolby Stereo productions through the encode/decode process so that any limitations of the process can be heard and corrected. This technique is well-established and routinely used in motion picture production today.

As a final note, the Dolby Stereo matrix process should be thought of as a technological art form that has been highly refined and well accepted as a means of creating more exciting and realistic soundtracks for audio/visual entertainment applications. As a consequence, decoding circuitry designed to reproduce such programs should not modify its technical characteristics, but rather emulate them. By so doing, the artistic content of the original production will be maintained during reproduction. On the other hand, signal processing techniques that make the system more discrete like and easier to produce with. are legitimate areas for improvement.

STEREO BROADCAST CONSIDERATIONS

There are a number of factors that influence the suitability of the matrix encode/decode process that has been described for the MTS broadcast system. These considerations can be summarized as follows:

1. Is the process adequately mono compatible?

2. Can the signal be handled in the production or broadcast plant without significant degradation?

3. Can the signal be transmitted and received via the MTS system without significant degradation?

The subject of mono compability can be addressed independent of the second and third issues. From a theoretical standpoint, one element of the process that is not mono compatible is a situation in which a signal has been encoded into the surround position. Since the resulting L_T and R_T signals would be equal in amplitude and opposite in polarity, they would sum to zero in a mono system. In practice, however, this is not a problem for several reasons. For the most part, there is never a reason to create an important audio effect in the surround channel only. If such an effect is, however, necessary, it is possible to pan the sound most of the way towards the surround location to produce the desired effect, the compromise being a reduced amplitude mono signal. Other uses of the surround channel translate very logically to mono reproduction. For example, back to front motional effects simply get louder as they move to the front and softer as they move to the back. From the standpoint of frontal sounds, the system is strongly mono compatible due to the fact that center images must be in-phase and of the same amplitude in order to be properly decoded. As a matter of historical note, one of the performance requirements of the Dolby Stereo matrix process was that of being mono compatible so that 35 mm two-channel optical prints could be successfully screened in mono equipped theaters.

As far as handling encoded program material in the production or broadcast plant is concerned, the situation is really no different than with conventional two-channel stereo material. Because of the common practice of discrete left and right signal routing, amplitude and phase integrity must be carefully monitored to assure a compatible mono sum signal.[7,8]

Looking at the transmission/reception link of the MTS system reveals that there are a number of factors that can impair proper reception of encoded program material. These problems are typically perceived as localization errors involving front channel leakage into the surround channel as well as program modulated hum and noise in the surround channel. A number of factors responsible for these problems are described in detail by Foster[9] and are expected to be reduced with improvements in transmitter and receiver design. In particular, incidental phase modulation (ICPM) of the video signal into the audio during both transmission and reception continues to be a problem along with left/right tracking errors in program limiting and compression.

THE ENHANCED MATRIX DECODER AS A BROADCAST TOOL

In order to evaluate broadcast plant and transmitter performance during Dolby Surround productions and transmissions, a decoder such as the one shown in Figures 6 and 7 has proven to be helpful in addition to phase scopes and sum and difference metering devices. This form of monitoring, configured as shown in Figure 8, provides the particular benefit of allowing one to hear transmission problems that cannot be easily detected by conventional two-speaker stereo monitoring techniques. In addition to monitoring matrix encoded transmissions, this form of listening is also useful in detecting subtle errors in conventional stereo transmissions. A common problem often encountered is the improper use of stereo synthesizers with automatic stereo detecting circuits. During long dialog program segments, the stereo detector determines that the program is mono and switches in the synthesis circuits. When a dominant stereo effect eventually comes on, it switches the synthesizer out. This cycling on and off can be rather subtle with casual two-speaker monitoring, but becomes quite obvious with the decoder scheme described due to the excessive surround information generated during stereo synthesis. Other difficulties such as missing channels, channel imbalance, and channel polarity flips are also more readily detected.

SUMMARY AND FUTURE TRENDS

If one considers television to have as one of its goals, the task of giving the viewer a sense of "you are there," the audio process described in this paper allows the broadcaster to take a significant step forward beyond two-speaker stereo. Just as the MTS stereo system is mono compatible, the matrix form of surround sound encoding described is stereo and mono compatible. This important compatibility, consequently, allows program producers to begin producing in the format now for future enhanced viewing, while at the same time satisfying todays' viewers. Even though high performance enhanced matrix decoders are just beginning to penetrate the consumer marketplace, advances in circuitry technology will give way to more cost effective designs that will eventually find their way into mass market consumer products.

To date, a number of experimental surround sound network broadcasts have been successfully completed involving movies, made-for-television productions, and live sports broadcasts. With additional broadcasts planned, the future looks promising for this use of the MTS and other two-channel television broadcast systems. In addition, future high definition television productions will also benefit from this process because of a greater demand for audio realism.

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Figure 6

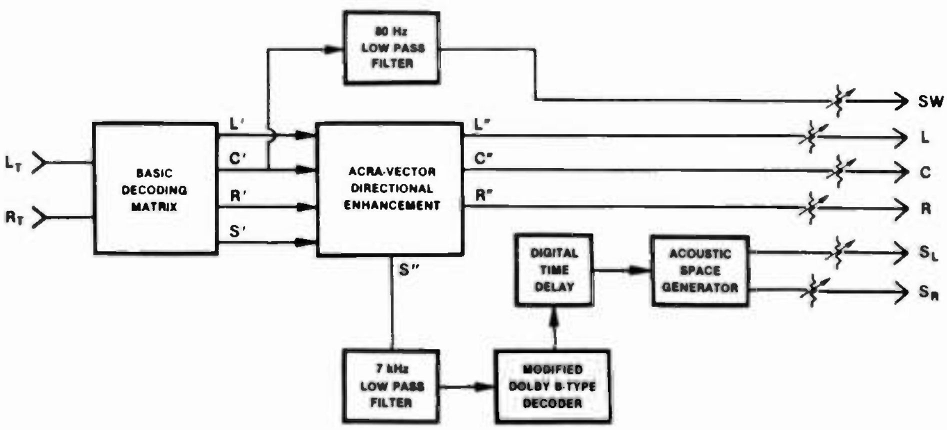

Figure 7

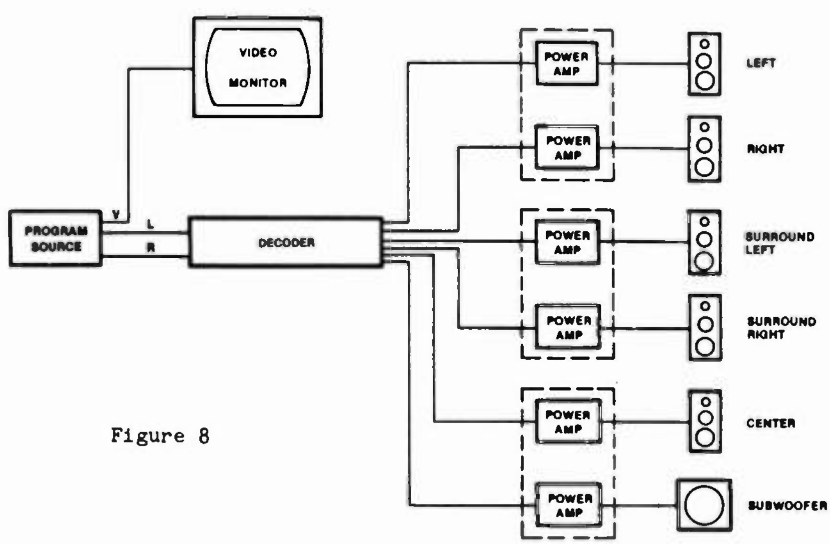

Figure 8

A STATE OF THE ART STEREO AUDIO PRODUCTION FACILITY FOR TELEVISION

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ABSTRACT:

CBS recently completed a program building facility at Television City meeting today's technical and production requirements, and allowing for the future needs of enhanced video and stereo audio. The facility provides for multi-machine video editing, complete stereo audio editing, audio sweetening, or full program coordination without hardware changes or modification.

CBS recently completed a program building facility at Television City that meets the technical and production requirements of today while allowing for the future needs of advanced video and stereo audio. This facility can accommodate:-

(i) Multi-machine video editing
(ii) Complete stereo audio editing,
(iii) Audio sweetening, or
(iv) Full program coordination.

All these are achieved without hardware modifications.

Parallel manual remote control and edit-computer control of all audio tape recorders, videotape recorders, and electronic character generators, and the audio/video switcher, was considered essential to minimize post-production time and cost, while making technical operation easier and more flexible. Both technical and production personnel needs were considered in detail to maximize the efficiency of the various operating configurations.

BACKGROUND

In 1979 CBS assembled a temporary facility to provide monophonic audio and graphic insertions into "spots" for the Promotion Department. This facility, known as PC 23, was assembled from various used and new pieces of equipment to meet the immediate demand. Over the next few years, outdated equipment was replaced with newer equipment when the promotional spot requirements became more exacting. For instance, in a typical twenty second spot, a live announcer, sound effects, and electronic graphics could be edited simultaneously using only an eight-track audio tape recorder slaved to the videotape recorder. Output from the old PC 23 reached approximately five thousand promotional spots per year, when it was taken out of service for a complete update and rebuilding.

NEW FACILITY CONCEPT

With the onset of stereo audio television broadcasting it was necessary to design the new facility to accommodate full stereo production. The high costs of the design and construction of a large control area with full stereo capability, prompted a design which could be used for multiple purposes. Hence the four-fold design concept was established incorporating:-

(i) Multi-machine video editing
(ii) Stereo audio editing
(iii) Audio sweetening
(iv) Program co-ordination

All audio tape recorders, videotape recorders, effects cartridge machines and an electronic character generator are controlled by a time code edit system to enhance the operations. In addition, complete on-line one-inch stereo videotape editing is provided using the same equipment.

Since a large amount of stereo audio equipment is installed, including a 24-track audio tape recorder, the room is also designed to do audio sweetening. External remote audio lines, video lines, and videotape recorder control facilities are provided between PC 23, the central videotape area, and the network transmission center. These facilities allow PC 23 to do on-air program coordination of prime-time specials and weekend sporting events.

PLANNING AND LAYOUT

Facility

The general layout of PC 23 was designed to follow that of a studio control room. In this format, the audio control area is separated from the video and production control area, which, in turn is separated from the videotape recorders and video/computer processing equipment (Figure 1). Each area is visible to the others by means of tinted glass sliding doors. A small announce booth is constructed such that the announcer has direct visual contact with the director and the audio mixer. This separate room layout significantly reduces ambient noise while enhancing the aesthetic qualities, and creating a more comfortable relaxed working environment. The segregation of audio, production, video/computer processing and electronics in a self-contained environment, also reduces the installation and debugging costs, and reduces future modification and upgrading costs. This layout facilitates maintenance because of the close proximity of all equipment in the system. In addition, the proximity of the videotape machines to the technical director and production staff, assists on-air program coordination by allowing visual confirmation that tapes have rolled, or if there has been a videotape machine problem.

The acoustical needs for PC 23 created special challenges. Even though the size of the new facility had been increased, the amount of equipment to be installed had increased even more. Paramount throughout all the planning and layout was the concern to keep the room symmetrical for stereo monitoring, in both the audio room and the production area. Given the fixed constraints created by the amount of equipment to be installed, it was found that certain designs would be more costly to construct without yielding any appreciable improvement in the overall acoustics. These problems were individually addressed and resolved in close colaboration with the Facilities Department.

Audio Configuration

The first consideration in the audio planning was the size, style and quantity of audio consoles. It was decided that rather than use one very large console requiring two audio mixers, a smaller main mixing console would be augmented by a separate effects console (Figure 2). An operator for the sound effects console would only be required in applications where cartridge sound effects were needed. The location of the main audio console was selected so that the operator and not the console was positioned in the center of the room for ideal stereo monitoring.

Because of the other uses of PC 23 previously described, it was necessary to install patchable ancillary audio equipment. The problems associated with the installation of this additional audio equipment and patch bay were resolved by the construction of a console that would be able to house all of these items as well as the audio effects cartridge machines. The effects console was placed behind the main console and as close to the center of the room as possible, enabling the effects mixer operator to utilize the main stereo monitoring.

A 24-track audio tape recorder was chosen to meet the needs of promotion production and audio sweetening. Two 1/4-inch audio tape machines were required for pre-recorded material. Machines were selected that were capable of providing a third time code track that would not interfere with stereo audio.

An external audio matrix was provided to allow for full computerized audio editing. Manual control of this matrix was installed in the production room to facilitate simple one-inch editing.

Audio monitoring for the audio room was provided by three large loudspeakers, two for stereo and one for monophonic audio, suitably positioned and angled for optimum stereo sound to the main mixer operator. The monophonic sound is generated by mixing the left and right program output of the room rather that taking a direct feed from the audio console. This allows the mixer to make a true judgement of the stereo quality by listening to the combined monophonic signal after all processing and distribution equipment. In addition to the main audio monitoring, a small monophonic speaker is provided for the effects console and audio tape operator for cueing purposes.

Video monitoring in the audio room is provided by a large color line monitor and several small black and white videotape playback monitors. The monitor array is located so that both the main console mixer and the effects console mixer have an unobstructed view. This array of monitors allows the audio mixers to visually anticipate their cues in addition to reacting to verbal directors' cues.

Video Production Configuration

The several configurations of PC23 require more video switcher inputs than would be used in a single operation. It was determined that it would be more cost effective to subswitch the video inputs based on the operation, rather than to purchase a larger switcher (Figure 3). Furthermore, the constraints created by the size of the production room limited the amount of space available for the switcher control panel. Certain switcher features were deemed essential for the various operations. These include definable wipe patterns, special effects recall through a programmable memory, full switcher configuration storage and recall, chroma key, borderline generation, and computer control.

Graphics are provided from a remote titling camera and an electronic character generator with its operator located in the production room.

The video production console required a compact layout in order to accommodate the video and editing control equipment, a technical director or editor, two production personnel, and a graphics operator. The main objective was to provide comfortable space for the technical and production staffs. In the equipment layout of the console, primary consideration was given to the edit position, since this operation would occupy 80 per cent of the room time. Secondary consideration was give to the technical director's position, while ample room was left at the end of the console for the graphics operator control panel and script reader.

Two color monitors were provided for line and preview for both production and technical staff. Black and white monitors of all the video switcher inputs were provided for the on-air program coordination operation. A test station was installed in the technical area of the production console for video timing and quality checks.

A color monitor was provided for the graphics operator to preview upcoming events. Small color monitors were placed on the production console for creation and composition of graphics. Finally, a television receiver was provided to check the quality of the program as it would appear on a home receiver.

The audio monitoring paralleled that in the audio room, with three speakers installed, two for stereo and one for a mono-mix. Again, the mono-mix was provided from the left and right program to ensure proper levels and phasing of the final stereo signal. The speakers were positioned and angled to give optimum stereo sound to the production person seated at the center of the console. In addition a pair of stereo near-field speakers were placed on the console to monitor the stereo in a format that might be more compatible with home receivers.

Control

The primary objective of the promotion, editing and audio sweetening operations was to have as much equipment as possible controlled by computer and referenced to time code. With this configuration, promotional spots or edits could be repeated with all inserted events occurring at the same time and with the ability to make precise time adjustments, if required.

Remote manual operation of the videotape machines was necessary for the on-air program coordination application, in order to accurately roll commercials at any given time. Also, control from the Network Transmission Center routing switcher was made available to route incoming live programming for coordination.

SYSTEM DESIGN

Audio System

The audio system is designed around a 32-input master recording and mixdown console with separate microphone and line input connections. Twenty-three of the playback lines from the 24-track audio tape recorder are attenuated and fed into the microphone inputs. The twenty-fourth microphone input is used for the announcer. The stereo outputs of the edit audio switcher are reentered into the audio console line inputs. In addition to the edit switcher, the other pairs of line inputs carry the six playback switcher outputs routed from the main videotape room, two one-inch videotape machines dedicated to PC23, one Umatic video cassette machine, three outputs from the Network Transmission Center routing switcher, two 2-track audio tape machines, four sub-master outputs from the effects console, and two reverb return lines. Additional pairs of monitoring inputs are used for edit preview, 2-track audio tape recorders, and the dedicated one-inch videotape recorders.

The 24-track channel outputs of the audio console which are assignable through a central assignment system, are returned to the record inputs of the 24-track audio tape recorder. Sixteen of these channel outputs are paralleled to feed the inputs of the audio edit switcher. A cue output is returned to the announce booth to provide a mix-minus program cue. Separate stereo outputs are used to feed the audio room and production room monitor systems. As was previously mentioned, these monitor outputs are electrically summed outside the console and fed to a third loudspeaker for monophonic monitoring. Two stereo outputs are fed to reverb units. The main output is distributed from external amplifiers to the two dedicated one-inch videotape machines, to the Network Transmission Center routing switcher, to the main videotape room record routing switcher, and to an external phase meter located in the production room console.

The effects console is a 16-input unit with four assignable sub-masters and one line output. The sixteen inputs are fed from ten mono, two stereo playback only, and one stereo record and playback continuous loop cartridge machines. The submaster outputs are fed directly to the main audio console, while the line output is used only for local cueing.

Time Code System

The time code system consists of two 6-input selectors. The first selector's inputs are from a time code generator, the two dedicated one-inch videotape machines, the Umatic video cassette machine, and two patchable inputs. The output of this selector is connected to the master input of a time code synchronizer that is used to slave the 24-track audio tape machine. The second selector's inputs are from the 2-track audio tape machines, the three videotape machines mentioned above, and a feed from the master time-of-day generator in the main videotape room. In addition to these facilities playback time code lines are routed to the audio patch bay for the six tape machine lines from the main videotape room.

Video System

The video system is designed around a 10-input switcher with one effects bus in addition to the program output. The inputs available to the switcher consist of the two dedicated one-inch videotape machines, six playback switcher outputs from the main videotape room, a feed from a titling camera, three outputs from the Network Transmission Center routing switcher, an electronic character generator, a test signal input, color black, and a background signal. The output of the video switcher is distributed to the two dedicated one-inch tape machines, the Umatic machine, the main videotape record routing switcher, the Network Transmission Center routing switcher, the audio room monitor, and the production room monitor selectors.

The two color monitors provided for the production area monitor different feeds depending upon the use of the room. To accommodate this, each monitor is fed from a six-input selector. The inputs of these selectors are identical and consist of program, preview, effects bus, edit preview, and two patchable.

A timing and video test station is installed in the outside equipment racks for initial setup and routine maintenance. This test station consists of a 20-input selector fed from all primary switcher inputs, the program, the preview output, and the effects bus output. The output of the selector is fed to a high quality vectorscope, waveform monitor, color picture monitor and SCH phase meter.

Control System

The control system is designed around the edit system. The two one-inch videotape machines, the Umatic cassette machine, the video switcher, and the audio switcher, are all interfaced to the edit system. In addition, control lines from the edit computer are connected to an existing control patch to enable additional one-inch videotape machines to be connected to the system. A time code synchronizer is utilized to slave the 24-track machine to the playback videotape machine.

Start, stop and record functions of the 24-track and the 2 two-track audio tape recorders are controlled by relay closures operated by the edit system. Similarly, the cartridge machines start/stop and the electronic character generator insertion are controlled by relay closures of the edit system. All of the controls are connected in parallel with the standard local controls of the equipment. In addition to the edit control of the cartridge machines, a set of start/stop push buttons is installed in the audio effects console for additional flexibility.

Videotape machine control for the on-air program coordination operation requires a separate set of manual remote controls, since not all technical directors are trained for, or comfortable with the edit keyboard operation for switching on-air programs. This control panel consists of individual start/stop pushbuttons for the six machine positions that can be routed from the main videotape room. In addition, two assignment switchers are available to control the machines in groups.

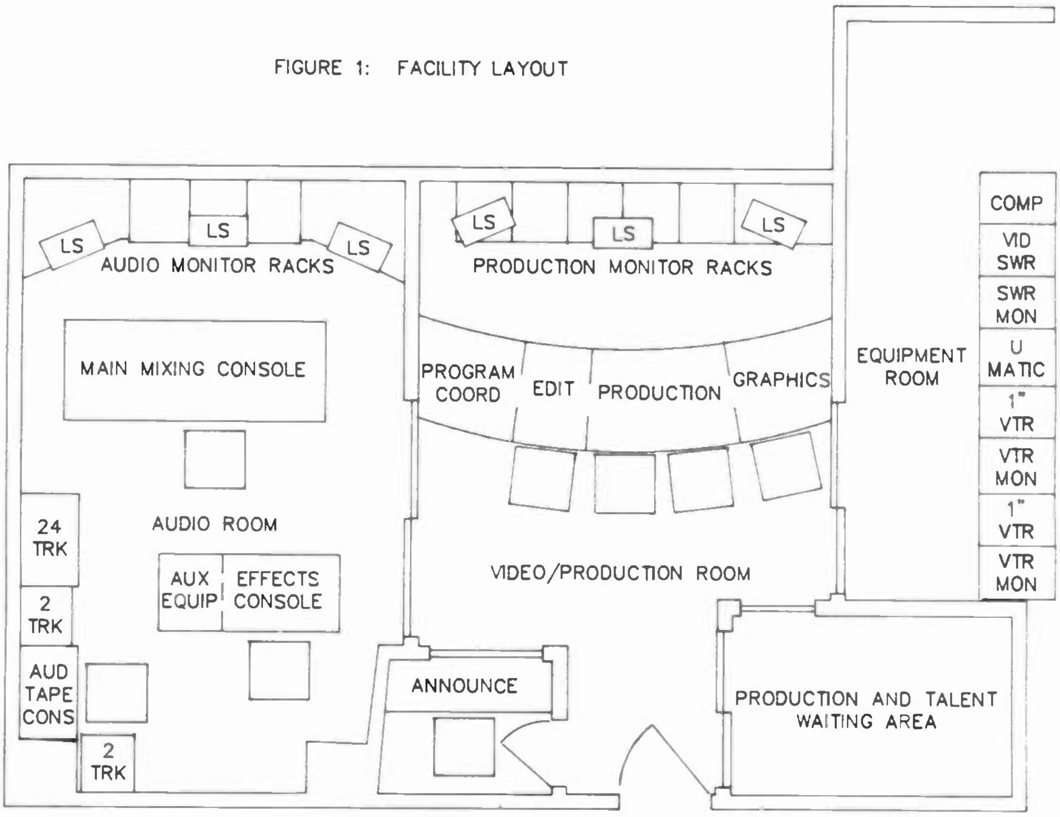
Future Considerations

Future upgrading of this facility, whether for digital audio, component video, digital video, or even high definition video should have construction and installation costs minimized because of the centralization of the equipment described. In any subsequent upgrade, the major costs will be for new equipment rather than for installation. Many future needs are already provided for through the flexibility and multi-purpose design of the facility. PC23 has already been used to transmit Dolby surround stereo audio for the CBS coverage of Super Bowl XXI.

Acknowledgements

The facilities described have resulted from the combined efforts of many people. The authors wish to thank in particular the members of the CBS Television Facilities Department, the Technical Operations Department and the CBS Engineering and Development Department whose efforts contributed to the success of this project.

FIGURE 1: FACILITY LAYOUT



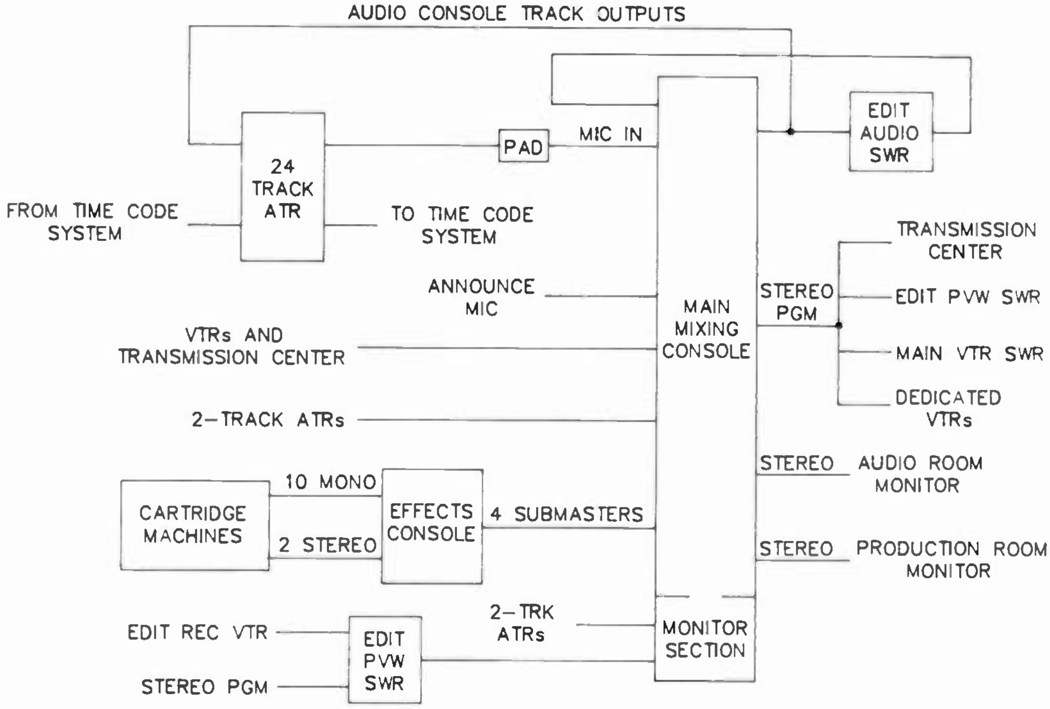


FIGURE 2: SIMPLIFIED AUDIO SYSTEM

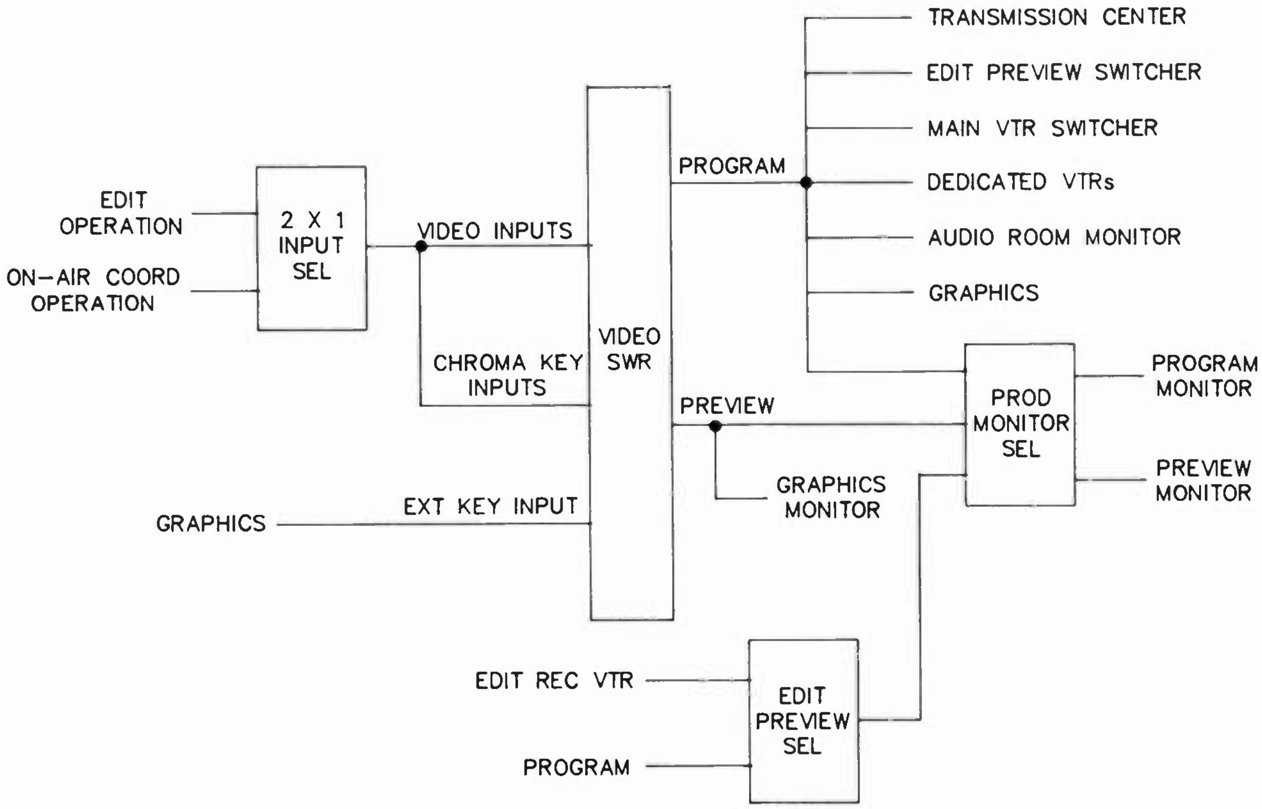


FIGURE 3: SIMPLIFIED VIDEO SYSTEM

DIGITAL SOUND AND DATA FOR BROADCAST TELEVISION—A COMPATIBLE SYSTEM

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Dolby Laboratories
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0. ABSTRACT

Laboratory tests have shown it possible to add an additional digitally modulated carrier to the broadcast television signal. A QPSK carrier modulated with 512,000 bits per second of information has been introduced at a frequency 4.85 MHz above the video carrier frequency. The carrier level is -20 dB with respect to the vision carrier. Tests show the new signal is compatible with the BTSC stereo and SAP signals.

I. INTRODUCTION

The BTSC stereo system adopted for multichannel tv sound in the US is theoretically capable of excellent performance. Unfortunately, the realities of intercarrier analog sound prevent this potential from being realized. The output of a BTSC decoder invariably contains numerous 'birdie' and 'buzz' products. BTSC stereo quality is inferior to all other common consumer audio formats: compact disc, compact cassette, FM radio, analog disc, VTR and video laser disc. At present, BTSC quality is not a major issue. After all, it is "Stereo". The adoption of a stereo tv system has stimulated improvements in all areas of tv sound production. More care is taken during the production of stereo audio and the use of audio noise reduction is becoming more common on the VTR. Eventually, the better quality source material and the quality conscious consumer will begin to meet in mass, and the BTSC stereo system will become a limitation to the broadcaster.

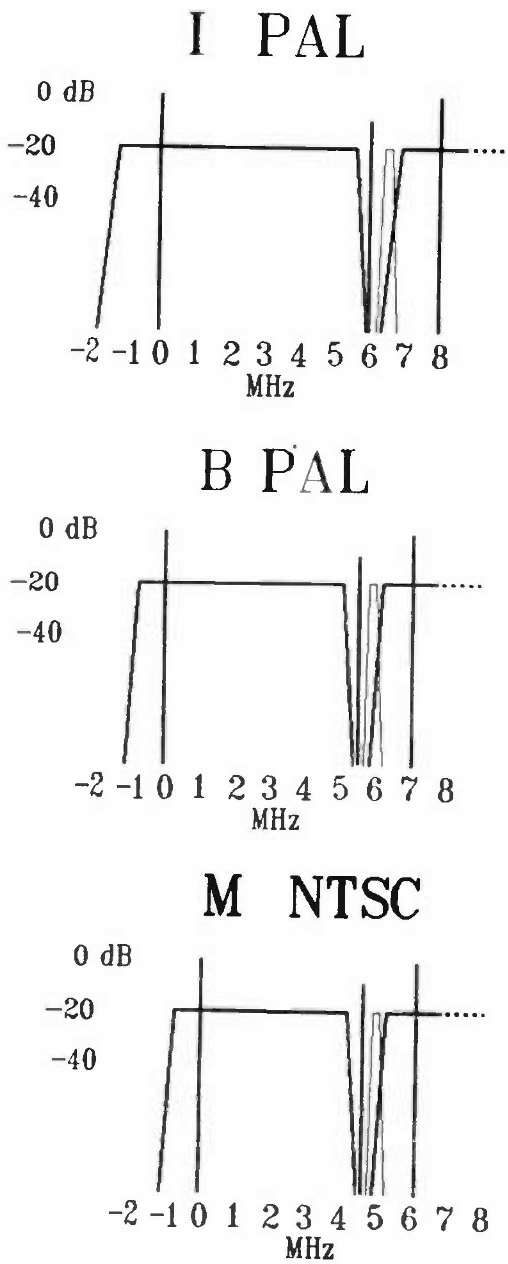
It would be very desireable to be able to add a digital stereo audio capability to the existing tv signal in a compatible way. Besides being able to deliver superb sound quality to their audience, during certain times of the day, broadcasters would be able to sell the data channel capacity to whoever has a use for it. This could become a major revenue source for broadcasters!

In 1983, the BBC conducted a series of tests at Wenvoe in South Wales of an experimental digital stereo sound system for television system I-PAL (Fig 1a).[1] The tests showed that the additional carrier was rugged and could be received even under adverse reception conditions. In 1984, tests conducted in London showed good compatibility with existing tv receivers. Further tests have since been conducted in Hong Kong (I-PAL), and in Stockholm and Helsinki with system B-PAL (Fig. 1b). The narrower bandwidth of the B-PAL broadcast signal makes it more difficult to insert the digital sound carrier without creating interference. In order to avoid interference the Scandinavian broadcasters were attracted to the Dolby adaptive delta-modulation audio coding method[2] which allows a 30% bandwidth reduction compared to a companded PCM system. Dolby Laboratories supplied sound coding and digital multiplexing equipment for broadcast tests in Hong Kong, Sweden, and Finland. While observing the work being done in Scandinavia, and comparing the B-PAL broadcast spectrum to that of M-NTSC (Fig. 1c), it became apparent that the results should be applicable here in North America. Both the B-PAL and M-NTSC systems have the same 750 kHz spacing between the FM sound carrier and the corner of the adjacent channel lower sideband. Work thus began to determine whether a digital carrier could be included into the M-NTSC broadcast signal without creating interference to either the existing vision or BTSC sound signals.

II. SYSTEM REQUIREMENTS

The primary requirement for the system is compatibility. The new carrier must not interfere with existing services. The system should work on cable, where adjacent channels are used. There is potential for interference into the BTSC stereo signal, and the adjacent channel

Fig. 1 RF Spectra of TV Systems showing additional QPSK data signal.



vision signal. The system should have no effect on BTSC stereo, and minimal effect on SAP. The BTSC stereo signal has a wider spectrum than the B-PAL FM mono sound signal so there is greater susceptibility to interference. The bandwidth of the new signal will have to be constrained to the minimum value which still allows solid error free performance in typical receive situations. This implies operation at the minimum possible data rate. A data rate of 512 kbit/s has been chosen. With the appropriate audio coding this rate is sufficient to broadcast two very high quality audio channels along with synchronization and mode signalling bits, and 64 kbits/s of auxiliary data.

Numerous digital modulation methods are available ranging from very simple and bandwidth inefficient to very complex and highly bandwidth efficient. There are inherent tradeoffs between efficiency, ruggedness and complexity. The best tradeoff between these factors is offered by differential quadrature phase-shift keying. Bandwidth efficiency can approach 2 bits/Hz. IC chips have been developed to demodulate QPSK and are available from at least two sources.

III. QPSK MODULATION

QPSK modulation may be thought of as a pair of bi-phase modulation systems operating in quadrature (fig. 2). Filtering is used to constrain the data spectrum. In order to transmit 512 kb/s, each path of the modulator handles 256 kb/s. The spectrum of a 256 kb/s data stream is shown in Figure 3a. the spectrum has nulls at 256 kHz, 512 kHz, etc. The spectrum extends to infinity. It is theoretically necessary to transmit only the lower 128 kHz of this spectrum. In practice it is necessary to send on the order of 200 kHz in a practical system. The filtering which is applied to the data must meet certain criteria for optimum performance. A parameter referred to as Alpha specifies the excess bandwidth over the minimum which is theoretically necessary. Experiments have been performed with filters corresponding to Alpha values of 0.3 (30% excess bandwidth) and 0.7 (70% excess bandwidth). The filtered baseband data spectra for both filters is shown in figure 3b.

Fig. 2
QPSK Modulation

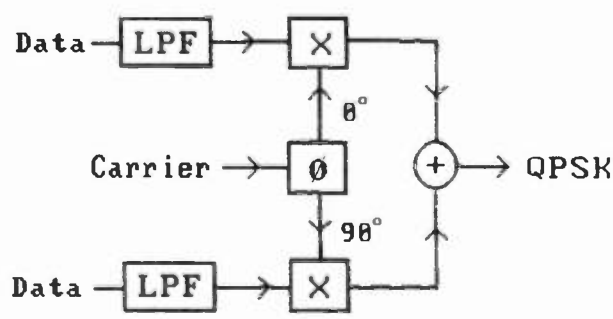
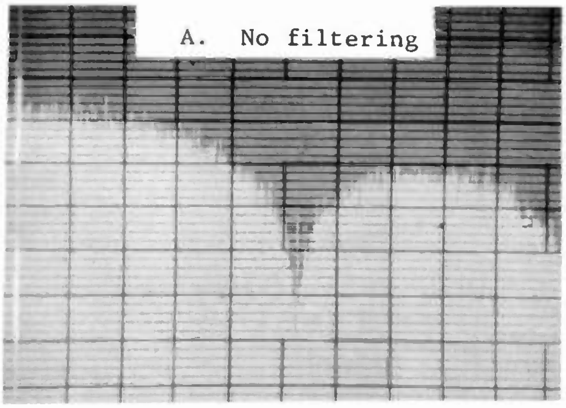
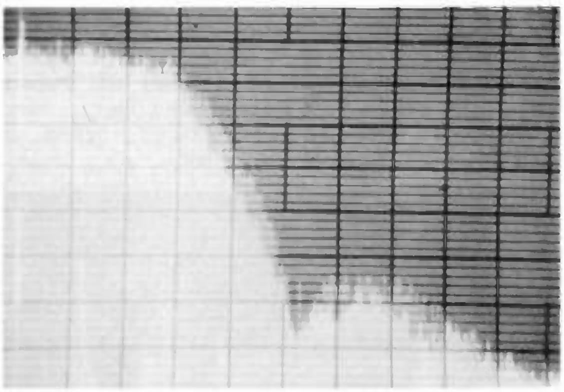


Fig. 3 Baseband data spectra of 256 kb/s psuedo-random data. Vertical scale 10 dB/div. Horizontal scale 50 kHz/div.

A. No filtering



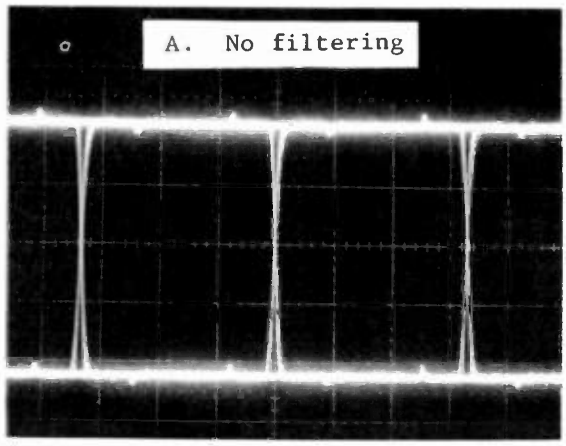
B. Alpha=0.7 (bright) and Alpha=0.3 filtering



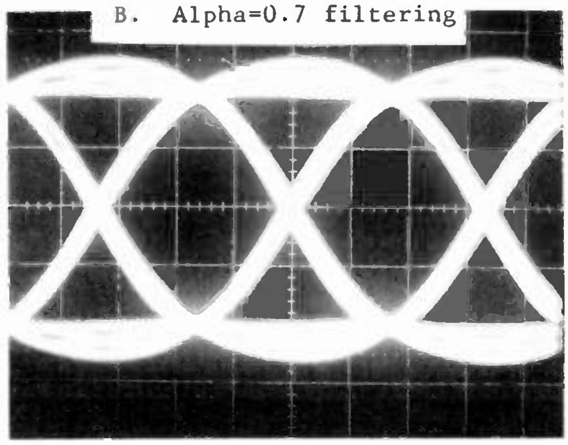
As the data is filtered, it changes in the time domain. This is shown by means of 'eye' diagrams in figure 4. This is what the data decoder looks at when it is trying to decode the incoming waveform into the original data pattern. The unfiltered data (fig. 4a) has a wide open eye and no data transition jitter. The Alpha=0.7 filtered data (fig 4b) has an eye only maximally open at one point in time, and some transition jitter. The Alpha=0.3 filtered data (fig 4c) has a much more complex pattern and lots of transition jitter. The complexity of the eye pattern correlates directly to the ruggedness of the data when passed through channels with imperfect amplitude flatness and non-linear phase response. The more complex eye pattern will degrade rapidly when exposed to any additional filtering, or effects such as multipath. This will reduce the ability of the decoder to tell 1's from 0's, and will result in data errors. For a consumer system, it is desireable to keep things simple and non-critical. The Alpha=0.7 filtered is thus preferred.

Fig. 4 Eye patterns at data demodulator.

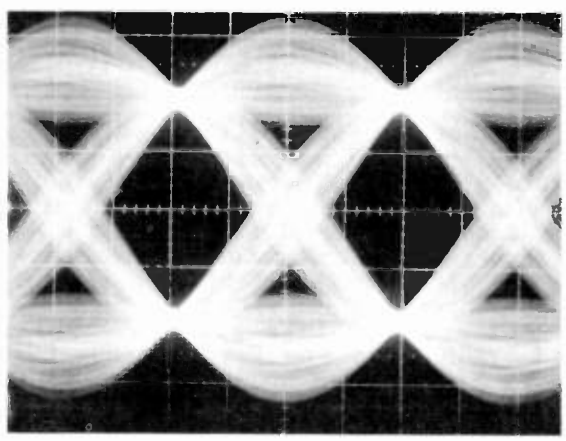
A. No filtering



B. Alpha=0.7 filtering



C. Alpha=0.3 filtering



IV. BTSC Compatibility Tests

The new signal can easily be made to have a noise-like spectrum independent of the naturally occurring data spectrum. This is accomplished by summing the data with a *pseudo*-random noise sequence. At the receiver, the original data is recovered by again summing with the same PN sequence. The effect of the new signal on the BTSC sound signal will be that of additive noise. The interference will be worst into the SAP channel since the SAP spectrum extends farthest away from the 4.5 MHz carrier (the 'Pro' channel is being neglected in this paper). Interference will be less severe into the L-R signal, and less again into the L+R signal. Testing for BTSC compatibility requires measurement of the additional noise caused by the new signal.

Figure 5 shows the test setup. A pair of high quality CATV modulators generate adjacent channels 3 and 4. A BTSC generator is used on Ch3, as is the QPSK modulator. The combined RF signals are fed through an RF attenuator to receivers. Tested receivers included a high end consumer tv set (Sony XBR) and the best quality BTSC decoder which could be found: the General Instruments (Jerrold) 'Starsound' BTSC receiver. Figure 6 shows the RF spectrum delivered to the receivers.

Due to the companding applied to the L-R and SAP channels, it is necessary to use a test signal to 'open up' the receive expander so that the noise will be apparent. The test signal chosen was 5 kHz at 100% modulation. In order to be most critical of the interference caused by the QPSK signal we want to obtain the best possible performance from the BTSC system. This is achieved by using the best receiver operating with plenty of RF signal level. The RF level for the tests was 27 mVrms. Figure 7 shows the SAP output spectrum of the Sony XBR with video (color bars) modulation on (upper trace) and off (lower trace). To give BTSC the best chance, compatibility measurements were done with no video modulation. Figure 8 shows the performance of the Sony XBR (upper trace) compared with the GI Starsound receiver (lower trace). The GI receiver uses a separate detector for the L-R and SAP signals and is able to achieve superior performance with this technique.[3] Since the GI receiver gives much better performance, it was used for the tests. Figure 9 shows what happens when the QPSK signal is turned on (upper trace) and off (lower trace). Since the distortion components still dominate it is not really feasible to measure noise with a SINAD (notch out the fundamental and read what's

Fig. 5 Test Setup

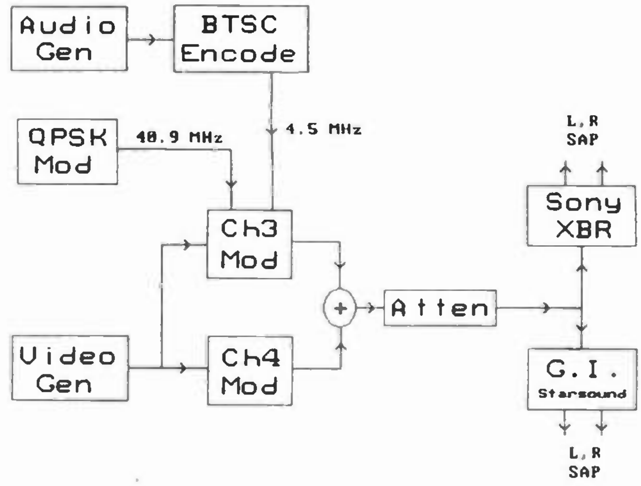
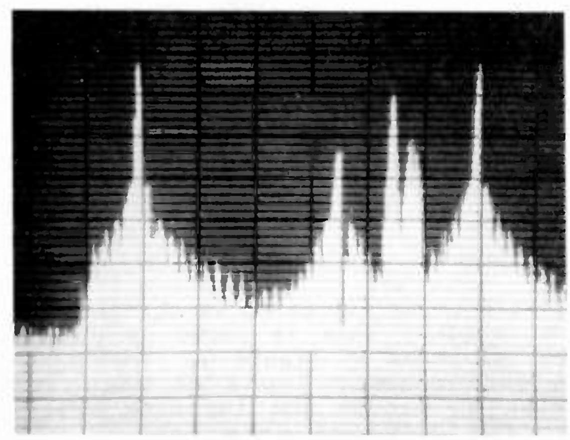


Fig. 6 RF spectrum of signal at receiver showing Ch3 vision carrier, color subcarrier, FM sound carrier, QPSK data carrier, Ch4 vision carrier.



left) measurement. With an HP3561A FFT based signal analyzer it is feasible to set up a band noise measurement. The vertical dotted lines in fig. 9 show the band 6200 Hz to 9600 Hz. Since there are no significant distortion products in this band, the total energy in the band can be used as a relative noise indicator.

To test for interference we measure the noise in the 6k2Hz to 9k6Hz band as a function of:

1) QPSK carrier freq. [4.80, 4.85, 4.90 MHz above vision carrier]
2) Data filtering applied. [Alpha = 0.7, 0.3]
3) QPSK carrier level. [swept down from -10dB rel vision carrier]

Fig. 7 Sony XBR SAP output. SAP modulated 100% with 5 kHz. Video modulated with color bars (top trace) and un-modulated (lower trace).

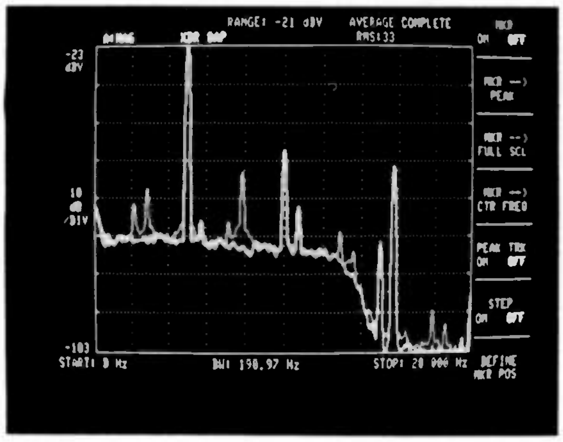


Fig. 8 SAP output of Sony XBR (upper trace) and GI Starsound (lower trace). SAP 100% modulated with 5 kHz. Video unmodulated.

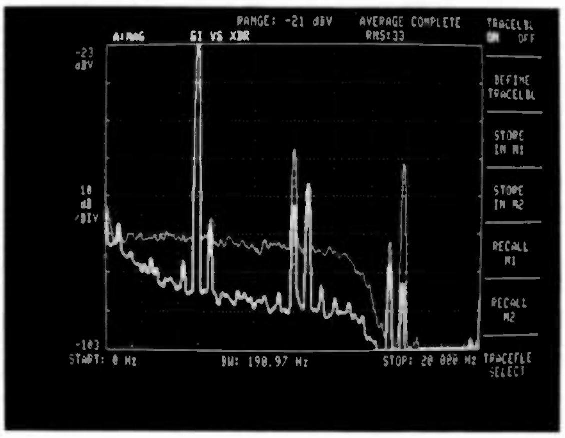
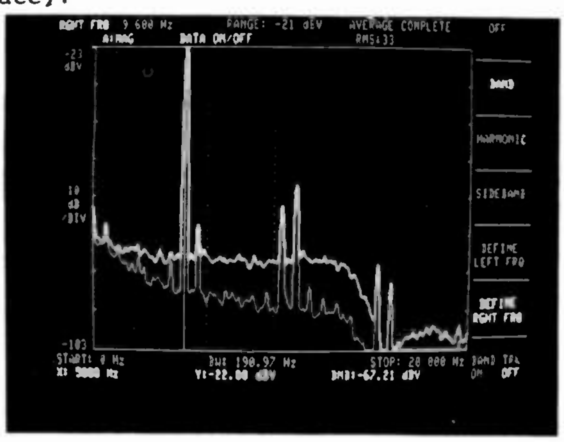


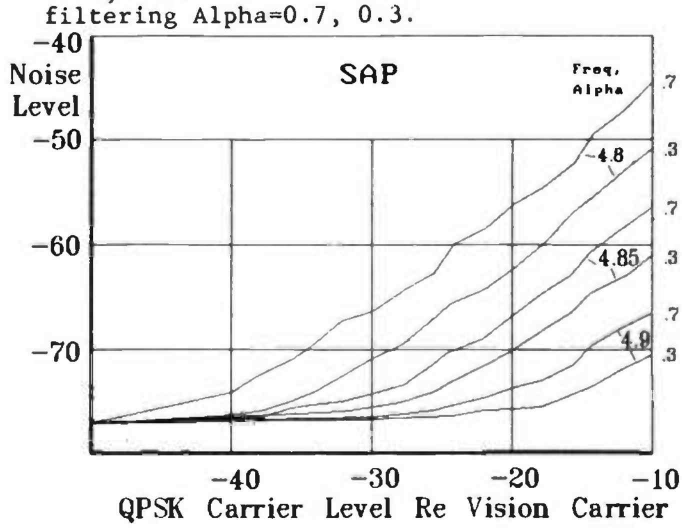
Fig. 9 SAP output of GI Starsound. SAP 100% modulated with 5 kHz. QPSK data signal on (upper trace) and off (lower trace).



The absolute numbers obtained are not of much significance. What we are looking for is the amount of noise increase caused by the presence of the QPSK signal, which will be a measure of the compatibility. If the QPSK signal increases the noise by only a little bit then it can be considered compatible. If it increases the noise by a great deal, then it would be considered incompatible. The judgement of what constitutes compatibility is a subjective one and different judges might come to different conclusions given the same data.

The results of the SAP tests are shown in figure 10. As expected the noise level is worst with the QPSK carrier closest to the FM sound carrier (4.8 MHz) and with the wider filter (Alpha=0.7). Using a narrower filter (Alpha=0.3) moves about half way to the 4.85 MHz Alpha=0.7 curve, which indicates that the extra filtering is narrowing the low side of the transmitted data spectrum by about 25 kHz. The difference between Alpha 0.7 and 0.3 filters would imply a narrowing of 50 kHz but since the filtering is split between the transmitter and receiver, the actual narrowing in the channel is less than might be expected. At the target operation point of 4.85 MHz, Alpha=0.7, -20 dB, the degradation to SAP is approximately 9 dB. If the carrier frequency is lowered, the interference rises rapidly so 4.85 MHz appears to be the closest the QPSK carrier can be placed to the 4.5 MHz sound carrier. Operation at 4.9 MHz would only create a few dB of degradation, but may create problems with adjacent channel operation.

Fig. 10 SAP noise level in band 6200 Hz to 9600 Hz. GI Starsound receiver. SAP 100% modulated with 5 kHz. Video unmodulated. QPSK modulated data at 512 kb/s. QPSK carrier frequency at 4.80, 4.85, 4.90 MHz above video carrier. Data filtering Alpha=0.7, 0.3.



A 9 dB noise penalty might be considered large, but it must be put in perspective. This noise penalty only occurs when:

1) A high RF signal level is used (20 mVRMS).
2) A special receiver is used (GI Starsound).
3) No video modulation is present.

Figure 11a shows the SAP with the QPSK on (4.85 MHz, -20 dB, Alpha=0.7) and with video modulation on (upper trace) and off (lower trace). Even with the noise made 9 dB worse, the output spectrum is still totally dominated by distortion products. This might lead one to question whether the extra noise is even audible. Figure 11b shows the same situation but with the video left on and the QPSK turned on (upper trace) and off (lower trace). The increased noise in the spectral holes between the distortion products is apparent and can be heard.

The penalty for the addition of the QPSK signal is a noise penalty in the SAP channel which is audible ideal conditions. Under practical conditions the noise change would probably not be noticeable.

Of more concern is the change in noise level in the stereo signal. This is shown in figure 12. While it is difficult to make out the individual curves at the bottom, the significant finding is that if the carrier frequency is 4.85 MHz or higher, there is no effect on the BTSC signal. At the target operating point, the interference caused less than 1 dB degradation in noise.

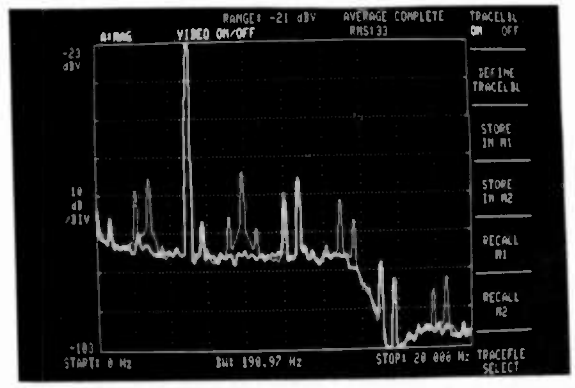
V. CONCLUSION

A careful test has shown that a digital carrier can be added to the broadcast television signal in a way which is compatible with the BTSC stereo audio system. Much work remains to be done. The effect of the digital signal on the picture has to be rigorously tested. Adjacent channels must not be interfered with or the system will not be usable on cable. Practical ways to transmit the signal must be devised. The system must be field tested both over-the-air and on cable. For broadcast use, regulatory approval must be obtained.

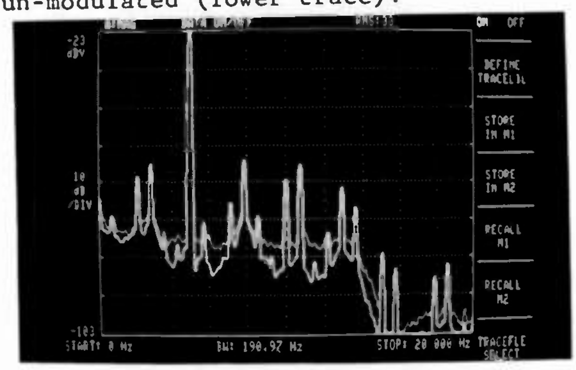
1. S. R. Ely, "Experimental Digital Stereo Sound with Terrestrial Television: field-tests from Wenvoe", October, 1983. BBC Research Dept. Report No. 1983/19.

2. Craig C. Todd, "Efficient Digital Audio Coding & Transmission Systems", NAB Proceedings, 39th Annual Broadcast Engineering Conference,1985.

Fig. 11 SAP output of GI Starsound receiver. SAP 100% modulated with 5 kHz.

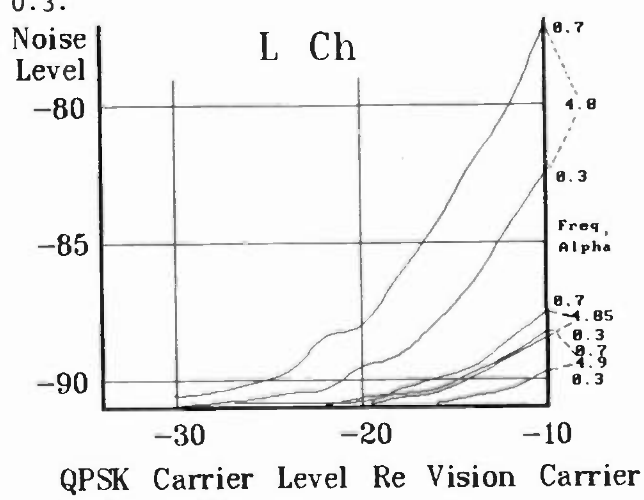


A. QPSK data on. Video modulated with color bars (top trace) and video un-modulated (lower trace).



B. Video modulated with color bars. QPSK data on (upper trace) and off (lower trace).

Fig. 12 Left channel noise level in band 6200 Hz to 9600 Hz. GI Starsound receiver. L channel 100% modulated with 5 kHz. Video un-modulated. QPSK carrier frequency at 4.80, 4.85, 4.90 MHz above video carrier. Data filtering Alpha=0.7, 0.3.



3. Clyde Robbins, "BTSC: The Stereo for Cable", 1986 NCTA Technical Papers, March 15, 1986.

STATE-OF-THE-ART GRAPHICS FACILITY

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B. Lilly
(Formerly with CBS Operations and Engineering)

ABSTRACT

This paper describes a state-of-the-art animation system used in the 1986 CBS News Election coverage. After presenting a brief history of Election graphics at CBS, the technical requirements, equipment selection, and system design of the facility are described in detail.

1. HISTORICAL BACKGROUND

Early Election graphics consisted of "boards" on which numerical data (corresponding to raw votes, and analyzed key precincts) were entered manually. These boards were located around the periphery of a studio set so that a camera could pick up data on any board at will and allow the correspondents to keep up with the latest changes.

In the early 1960's an improvement was effected by the use of electro-mechanical boards with remotely settable numerical displays. These were shown on-air by a camera as with the manual boards.

Starting in 1970 there was a desire to use the computer facilities then in CBS to display analyzed vote data directly. A modified IBM terminal displayed larger than normal characters which was then shot with a studio camera. About one half of the analyzed vote displays were done this way.

With the advent of electronic character generators, numerical data could be supplied and displayed electronically. The character generators were driven by external computers in place of the keyboards normally used. A potential source of error, viz. human error in transcribing data, was thereby reduced considerably. The backgrounds for these displays were art cards shown via a group of studio cameras.

Electronic still stores allowed prepared backgrounds to be shot and stored ahead of time and subsequently retrieved electronically, reducing the studio floor space and camera requirements.[1] A computer interface allowed rapid, random access to the appropriate background for a given race. Text continued to be added with computer controlled character generators. Alignment of text with the background was simplified in this all-electronic system.

1980 provided the first opportunity at CBS to air in-house computer graphics. An attempt to interface an Ampex AVA system to the election computers proved to be too difficult. The unit was used manually with limited success.

In 1982, it was desired to increase the impact of the graphic presentation through the use of animation rather than static displays. The Dubner CBG was used to provide animation. Use was made of the programmable nature of the CBG, which displayed animation sequences when instructed by the computer system. These animations were used for summary displays only. Displays of numerical data was still done with character generators and still stores.

In 1984 the Dubner CBG provided animated backgrounds for the numerical data as well as summaries. The unit also controlled an external selector to cause text to be keyed over the animation at the appropriate time. Keying was performed in the studio production switcher.

2. GRAPHICS REQUIREMENTS

It was desired by CBS News that the graphics displays for 1986 be of a very high quality. For this reason, it was not possible to use conventional graphics devices due to the simultaneous requirement for real-time playback of the animation. Currently available graphics generation devices are not capable of generating real-time animation at the level of image complexity desired.

Although the programmable nature of the Dubner CBG had been used to advantage, the software development was time-consuming and the graphics were not of the quality desired. A device was needed which would require a minimum amount of programming or adaptation to provide the animation required.

These requirements led to the choice of a record/playback device used with graphics which were prepared ahead of time.

3. GRAPHICS PRODUCTION

General animated sequences were prepared on a high speed, real-time, animation system. Specific information for each election race was added at a post-production house. This process generated a 1 inch type C video tape.

4. SYSTEM REQUIREMENTS

A graphics animation system for near-real-time presentation of graphics support for hard news events should meet the following requirements:

1. Ability to be automated -- computer supplied data is faster and more reliable than manual input. A minimum of human intervention is required.

2. Random access to certain portions of animation sequences is required. This is due to the fact that one does not have *a priori* knowledge of election winners or presentation order.

3. Equipment must be compatible with industry standards and practices. i.e. broadcast-quality video, genlock capability, etc.

4. Sufficient material must be readily available to provide animation supporting a broadcast which runs for approximately six hours[2].

5. EQUIPMENT SELECTION

The following equipment was selected as the basis of the 1986 Election graphics system:

1. Abekas A62 Digital Disk Recorders were selected as the animated graphics record/playback devices. This unit can store up to 100 seconds of high quality NTSC video. Random access capability is provided. Additional features aid in presenting smooth animation including computer control. Four A62's were used to provide simultaneous access to different types of animation sequences. One spare unit was available in the event of a failure.

2. Chyron 4100 character generators provided most of the variable text and numbers in the elections displays. These are the systems in daily use at CBS.

3. Video insert keyers (one per Chyron/Abekas combination) were used to key the text from the Chyron over the animation background played from the Abekas A62. Keying of text was performed upstream of the studio production switchers. Use of stand-alone keyers allowed the keyers in the production switchers to be used for other purposes, and decreased the number of switcher inputs which would otherwise have been required. A reduction of operating controls also resulted.

4. Sony BVH-1100A VTR's and BVE-1000 edit systems were used to transfer the source material from 1 inch type C video tape to the Abekas A62. A relay closure provided by the BVE-1000 was used to trigger the start of recording of the A62.

5. Custom-built interfaces between the Abekas A62's and the Chyron character generators were used to synchronize text displays with animation graphics.

The above equipment was installed utilizing existing plant facilities. This provided some cost savings in the installation and additional flexibility in the event of equipment failure. Simple patch cross repatching replaced a defective unit with a spare.

In addition to the broadcast equipment and interfaces enumerated above, a network of computers was used (as in previous years) to fulfill two objectives:

1. Provide up-to-date information to editorial decision-makers and on-air correspondents based on the latest election return and survey data.

2. Provide a system for controlling the broadcast graphics equipment.

Two computer systems, each with backup capabilities (redundancy), were used, each one dedicated to one of the aforementioned tasks. The systems were connected by communications lines.

6. ACKNOWLEDGEMENTS

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SORTING OUT GRAPHICS SYSTEMS FOR BROADCASTERS

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Electronic graphics equipment is by now ubiquitous in broadcast television plants. An overview is given of the basic types of electronic graphic systems. What follows is a discussion of management, software, and technical issues encountered in planning for the acquisition of equipment used in in-house electronic graphic systems.

TYPES OF ELECTRONIC GRAPHIC SYSTEMS

There are four basic classifications of electronic graphics equipment used in broadcast television production. These are character generators, electronic still stores, electronic paint systems, and 3D modelling and animation systems. Electronic graphics equipment may function in one or more of these basic categories.

Character Generators

Character Generators are used primarily for producing text or captions. Character generators were the first major type of electronic graphics equipment to be used in broadcast television, and are today an indispensable part of any broadcast facility.

Fonts or typefaces for character generators are generally loaded into the system from disk drives. In some character generators, the font information is carried on programmable read only memory devices (PROMS). The font data itself is usually represented by run-length-encoded information. The effective horizontal resolution of the character generator depends on the increments in which the duration of run-length segments can be varied.

Character generators differ in their handling of fonts, depending on how the system is designed. In some systems, the choice of on-line fonts must be specified when the system is booted. Other systems use software to dynamically load fonts into memory as they are needed to construct pages of text.

In some systems, the size of characters can be changed instantly, while other systems use font utility software to make a scaled font from an original. Font compose software allows the creation and manipulation of font data by the user.

The addition of edging enhances readability when the text is keyed over video. The strategy for putting borders and shadows on characters differs from manufacturer to manufacturer. Some units use hardware to add edging, while other units incorporate borders and shadows into the font itself. Or, a combination of hardware and software edging may be used.

Similarly, the antialiasing of text in character generators has been approached by the designers in both hardware and software.

Character generator text planes are usually displayed as horizontal rows of characters. Some manufacturers have gone to great lengths to provide more freedom of text layout on the screen by allowing characters and rows to overlap, or by eliminating row restrictions entirely.

Character generators usually employ a color palette. The palette contains all the colors which can be displayed at one time on the screen. Palette colors are chosen from a range of allowable colors which is determined by the hardware design, a number which can range from 64 to over 16 million possible colors.

Most character generators incorporate a background plane which can be made to appear behind the text. The complexity of the background can range from simple horizontal stripes, or colored boxes, to full-color pictures created on electronic paint systems.

One commonly available set of features are real-time effects. Such effects can range from rolls, crawls, and wipes, to real-time manipulation of characters in simulated 3D space. Other character generators compute animated effects a frame at a time, playing back the animation in real-time.

To aid the user in the execution of complex sequences during live production, character generator designers have provided automated modes of operation. Automatic sequencing features can include keyboard macros, event lists, and keystroke languages incorporating program constructs such as loops and variables.

Electronic Still Stores

Still Storage systems were the second major type of electronic graphics equipment to be widely applied by broadcasters.

In a still store, video images are digitized and held in a frame buffer. The frame buffer data is clocked out to a digital-to-analog converter, which reconstructs the original video. The data in the frame buffer is written to, and read from a magnetic disk for permanent storage.

Still stores have become more practical as the technology of disk drives has improved. Small Winchester drives are now available which can store hundreds of frames of video. This would have required the use of a computer mainframe-type drive only a few short years ago.

Multiple frame buffers in still stores provide two main advantages. They allow for multiple video outputs, or effects transitions between stills. Most all still stores incorporate a sequencing mode which allows the user to program random sequences of stills.

Some still stores incorporate indexing data with each still, and software to help keep track of the images. This is particularly important where the system allows the networking of multiple mainframes and disk drives, and the number of on-line stills is high.

More than just an electronic version of a slide projector, most still stores incorporate features for the compositing of layers for the creation of graphics. Usually, an outboard production switcher is used. If the layering process requires a digital-to-analog conversion, the number of layers is limited due to generation loss.

Electronic Paint Systems

Electronic paint systems allow the creation and manipulation of still images by means of software that simulates the tools of the artist. The user interacts with the system to manipulate the image on the screen through a bitpad or digitizing tablet. Software interprets this input and modifies the active image in random-access-memory (RAM), while displaying the result as a video output. Images are stored on a magnetic disk drive for later retrieval.

A variety of brush types are simulated in software including solid, airbrush, and wash, to name a few. Cut-and-paste, stencil, and typography are also common features in paint systems.

Some systems can display all possible colors at once. So-called "32 bit" systems represent each pixel with 8 bits of data for each of Red, Green, Blue, and Transparency components. Other systems use a color palette, displaying some fixed number of colors at any given time.

The type of frame buffer in which the video image is represented depends upon the system design. While most electronic paint systems use component storage of images, it is also practical to use a composite frame buffer as long as all the fields of a color frame sequence are stored.

Some electronic paint systems offer limited real-time effects, in the form of transitions between images, color cycling, and cel animation.

3D Rendering Systems

Three-dimensional modelling and animation systems allow objects to be specified mathematically, and then rendered on the screen from any simulated viewpoint.

Most 3D rendering systems allow for texture mapping, a technique whereby a two-dimensional pattern or image is mapped onto the three dimensional objects being modelled, much as wrapping paper on a gift box. Another common software feature is the use of simulated sources of illumination and surface reflectance characteristics to create light source shading.

The hardware for these systems ranges in complexity from personal computer (PC) based systems, custom designs, minicomputers, to supercomputers. The more sophisticated the computer, the faster objects can be rendered. Because of their relatively low cost, the PC-based systems in particular have found themselves applied in broadcast plants.

Because of the large amount of computation time necessary to render each frame, 3D modelling systems generally do not produce real-time animation. Instead, individual frames or small groups of frames are rendered and edited in sequence to videotape, under software control.

The predominance of conventional videotape as the output medium is being challenged by the advent of digital video disk recorders and tape machines. In their initial applications, the disk recorders were used as a direct replacement for videotape, and therefore received a composite video input from the rendering system. However, as digital video standards are developed, the trend is toward the transfer of video frames in the form of digital data to disk recorders and digital tape machines.

Other Types

Other types of electronic graphics equipment are used, which may have features of each of the basic types mentioned above.

Computer weather graphic systems have virtually eliminated hardcopy maps for displaying weather information. Weather graphics combine on-line data acquisition with a paint system, animation, and sequencing capability.

In teletext transmission, graphics are composed using paint system and character generator techniques. However, instead of transmitting video representations of the graphics, only the data necessary to reconstruct the graphics in a subscriber terminal is transmitted in the vertical interval of video.

ACQUIRING EQUIPMENT FOR IN-HOUSE ELECTRONIC GRAPHICS OPERATIONS

Management Issues

Ultimately, the application of electronic graphics techniques in the broadcast facility should fulfill the goals or expectations set by management. The mix of electronic graphic equipment in use at a broadcast facility should be considered as part of a system which is implemented to achieve these goals. The system consists of the equipment itself, the graphics generated by the equipment, the facilities scheduling, and the talented people who operate the equipment.

Management should consider the following issues when planning to obtain electronic graphic equipment:

Consistency. The marketing plan of a television station usually calls for the use of a unified design theme for all station graphics.

The choice of electronic graphics equipment should be influenced by the need to implement a design theme. To achieve consistency, this means that the equipment should be able to utilize a "toolbox" of elements which are designed to implement the look of the station. The design should be the primary mover in the appearance of the graphics, not the equipment itself.

Without providing this framework of predetermined elements, the bewildering array of graphic options available in modern equipment can slow down a production, or result in inconsistency as graphic solutions are created on the fly.

The use of toolbox elements for routine operations frees the users to apply creativty to projects where it is really needed.

Cost-Justification. Electronic graphic systems usually represent a significant investment of capital funds, and an operating cost which is ongoing. If additional personnel are going to be required to make use of the system, this is also part of the operating cost. Is the system going to be cost-justified? This depends on the application.

If there are direct revenues from in-house graphics production for outside clients, and/or if the new system reduces operating costs, it may be possible to project cost justification.

Where the primary goal of the system is to support the news and promotion efforts of the station, it is difficult to demonstrate cost-justification. The cost justification is presumably linked to the possibility of increased ratings and revenues, and is therefore difficult to prove. In this case, management must make the judgement that graphics systems are worth pursuing because improved graphics will contribute to the quality of the product and the success of the marketing plan.

Realistic Expectations. Is the proposed usage of the equipment going to be realistic? For example, weather graphic systems have improved to the point where one is tempted to use them for all sorts of production applications. But just try prying the equipment away from the weather person!

Or, how much use are you going to get from a paint system which is an add-on to the character generator which is already booked-up 20 hours a day for production and news supers? Maybe a stand-alone paint system is a better answer.

The moral of the story is, "Don't put all your eggs in one basket." Separate equipment usually provides more redundancy and scheduling flexibility. Also, the plant is less likely to come to a grinding halt if one piece of equipment fails.

Changing Roles. Are there any union / jurisdictional issues that will be introduced by the new technology? The application of electronic graphics equipment tends to challenge traditional operator roles. Management should plan for the need to have creative people interface with technical equipment. No one wants to implement a system in such a way as to precipitate a dispute.

Operator Input. It is always a good idea to involve the prospective users of the equipment in the evaluation process. Listen to their input. A small investment in travel costs to visit the manufacturer, trade show, or another facility can reduce the likelihood of wrong equipment being purchased.

Software Issues

Certainly, the supplier's parts and labor warranty are important considerations in specifying equipment. However, electronic graphic devices are essentially computers, and computers run on software. Most new features that are introduced for a particular graphics system are implemented in software. Therefore, it is critical to determine what the manufacturer's policy is on software and software updates.

Software Updates. How often does the manufacturer plan on issuing software updates?

Customer Feedback. What is the manufacturer's track record for listening to the end user when deciding what to put into new software? Suppliers who isolate their programmers from the end-user usually respond more slowly to customer input.

More Features. Are future software updates likely to result in more capabilities or easier operation of the equipment? Software usually takes longer to develop than hardware. In order to meet competitve deadlines when graphics equipment is introduced, preliminary software lacking features may be supplied. The buyer must be careful to know which features will come first, and which will be added later through software updates.

Software Charges. What is the manufacturer's policy on charging for software updates? Automatic software updates are frequently included in the purchase price during the warranty period. Beyond this period, there are two common policies on updates. Some manufacturers provide a software service contract which provides updates during the period covered by the agreement. Others charge a fee for each update. When a completely new feature is implemented in software, an extra charge could apply. Software updates should be considered as part of the operating cost of the equipment.

Manufacturers also have differing policies on how they correct outright bugs discovered in existing software which is out of warranty. One frequently-encountered problem is the practice of not correcting known bugs until the next scheduled release, unless individual users call and complain.

Technical Issues

The integration of electronic graphics equipment into the television plant raises some technical issues:

Key Signals. In addition to adding video sources to the plant, most graphics equipment also produces key signals, which are used by production switchers to "cut holes" for characters or graphics. Providing for a multitude of key signals in a moderately-sized plant might mean constructing an additional level of signal routing.

To further complicate matters, an increasing amount of graphics equipment generates linear key signals. These so-called "Soft Key" signals are necessary for the proper compositing antialiased graphics onto a video background. Some production switchers can be adapted to handle linear keys, and some cannot. Linear keys should be taken into consideration when specifying new production switchers.

Maintenance. Most of the currently available electronic graphics equipment is very reliable in terms of hardware.

The exception to this would be disk drives, keyboards, and digitizing tablets; which take mechanical abuse and are failure prone by nature.

But aside from these devices, the graphics hardware itself is usually so reliable that the maintenance staff might never have the chance to become familiar with the inner workings of the equipment by the time a failure does occur.

Maintenance training courses offered by the manufacturer are of some benefit. However, the best defense against excessive down-time due to equipment failure is an excellent customer service department, and a liberal board-swap policy. The manufacturer's customer service track record should be considered when acquiring equipment.

Computer Interface. Does the character generator interface to external host computers, such as election or newsroom automation systems?

Connectivity. Where different types of electronic graphic equipment are used to implement a design, it helps to be able to transfer elements between the equipment. Hopefully this transfer is in the form of digital data, avoiding the problem of generation loss. At present, one way of approaching this is to stick with a particular manufacturer if a means of exchanging image data between systems is provided. As standards for the exchange of digital image data are implemented, connectivity between dissimilar systems will hopefully improve.

CONCLUSION

Electronic graphics equipment has contributed greatly to the operation of broadcast facilities. These systems have, over recent years, gone from being a nicety to an absolute necessity in day-to-day production.

It is the hope of the author that the issues touched-upon in this paper will be of use in sorting out the various options in electronic graphics.

BROADCAST METEOROLOGY—A LOOK AHEAD AT DEVELOPMENTS IN WEATHER GRAPHICS DATABASES AND DELIVERY SYSTEMS OF THE NEAR FUTURE

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Television viewers and radio listeners in the foreseeable future will be the beneficiaries of new technologies in broadcast meteorology. The new technology will provide improved short-term local weather forecasting through advanced data collection systems, high speed communications, and mesoscale forecast modeling. Weather graphics will portray current data and forecasts of local weather events in high resolution and animation. This will be made possible through a new generate of government computers and processing called AWIPS-90 which will include advanced Doppler radars and communications at very high baud rates. Increased viewer and listener awareness of improved local weather forecasts, warnings and graphics will heighten competition for news ratings. Private weather companies will be the leaders in bringing this new technology to broadcasters and some of these new trends will be discussed.

Weather presentations on television evolved slowly in the 1950's and 60's accelerated a bit in the 70's, but in 1980 the old fashion weather board remained the main backdrop for presenting national, regional and local weather. Two graphics appeared in the 70's. One, a black and white satellite photograph and the other a color radar display.

The black and white satellite allowed viewers to see, for the first time, the intricacies and complexities of the atmosphere as shown by the detail of cloud photographs. Likewise, local radar pictures showed the complexity and detail of local precipitation patterns but the major changes in weather graphics and displays have come in the last 6 years.

The spectacular developments that have occurred since 1980 in the presentation of weather information and forecasts include the development, refinement and sophistication of color graphics equipment displays and the application of ingenuity and creativity to develop dozens of new ways to portray and display the weather and to peak viewer interest.

Simply by reviewing the tremendous changes that have occurred in the graphic offerings from Accu-Weather, we can appreciate just how much more sophisticated the television viewer has become in his understanding of weather phenomenon, weather patterns and how much more he is looking for; and likewise, how much the sophistication of news directors, general managers and weathercasters has increased just over the past few years.

When we began offering graphics four years ago, they were rather limited but yet they more than satisfied the wants and needs of television weathercasters, meteorologists and news directors. Now, we offer 800 graphics every day and the number and variety is increasing weekly. Let me give you some examples.

We initially offered national radars every hour, which are composites of over 100 individual radar scopes across the U.S. Then, clients indicated they wanted regional radars. As a result, we divided the country into six regions and began offering, in addition to the national radar composite, six regional color radar composites every hour. As some of you may know there are gaps in the national radar coverage because of insufficient radar sites and attenuation of the radar signal in certain areas because of terrain shadowing. As a result, there are many areas of the country where the radar representation is unreliable, inaccurate, and in some cases there simply is never any precipitation shown on national weather service reports. In order to overcome this, we have just completed development of an enhanced radar which we have called Newrad. Basically, we have had our computers decode and integrate more than 1000 hourly surface reports from around the United States and through a number of proprietary software packages, we have been able to integrate these surface weather observations into the hourly radar displays to show a more accurate precipitation coverage and intensity representation. We find that 5 or 10 years ago nobody would have cared, but today increasing numbers of clients have told us they will want to use the Newrad graphic, which will be demonstrated at the NAB for the first time. This new product will accurately show the approach and arrival of precipitation by looping. Likewise, some of our specialized products such as "Today's Worst Weather" our daily almanac feature and trivia question are being used to broaden the impact of weather to times outside of the traditional weather segment. Many of our stations are using these features in other parts of the news, that is as transitional slides, or as graphics for major news events where weather plays an important role. For example, we sold thousands of special graphics during the Chernobyl nuclear accident. We have had similar responses to special graphics documenting floods, hurricanes, tornadoes, and droughts.

We also see a definite trend toward more weather presentation. That is not to say that the length of weather segments are increasing. Rather more television stations want to be able to deliver 15, 30 and 60 second weather shows at times when meteorologists are not normally available. Also stations that have traditionally not presented weather are now looking for ways to do brief weather forecasts and even weather synopses. It is because of this demand that WeatherShow™ and WeatherBreak™ have been born in the past year at Accu-Weather.

Both are complete self-contained weather segments combining Accu-Weather graphics with voice-over audio delivery by a professional Accu-Weather meteorologist. The result is a professionally done weather segment complete with a highly accurate local forecast.

The WeatherShow™ is usually a 30 to 60 second feed with weather graphics displayed in a programmed sequence and synchronized with a recorded voice-over forecast by an experienced Accu-Weather broadcast meteorologist. Longer feeds are also available.

WeatherBreak™ is the same concept, only a much shorter segment lasting about 15 seconds, usually aired as a drop-in segment of a news break. "They can be used in stand-alone fashion as the weather news for brief updates or for in-depth presentations. The entire weather programming can be managed at low cost and little personnel time commitment at the stations. Once the system is set up, only a few minutes of an engineer's time is required each day to have complete weather segments ready to air.

These are not syndicated products; no two are alike. We are providing the service to stations in many markets, even to stations who on-air weathercasters for their main news broadcasts. In fact, we can and do work closely with a station's meteorologist to coordinate style, format, graphics and delivery."

We doubt this trend toward increasing weather presentations and increasing sophistication of weather displays is about to end. In fact, if anything, we believe it will accelerate. The reason is twofold:

In the United States, weather impacts so many activities and with business schedules getting tighter and tighter and with more people travelling every year, thereby being sensitive to the vagaries of weather, the interest is heightened and awareness of weather patterns and interest in weather forecasts in other areas is increasing. Secondly, developments now occurring within the government's National Weather Service are going to lead to a tremendous explosion in the availability and variety of weather data and information. The AWIPS-90 program which stands for the Advanced Weather Interactive Processing System for the 1990's will bring about this revolution. The specifications for this new system have already been issued, and this billion dollar project will become operational in the early 1990's. This new technology will provide improved short-term local weather forecasting through advanced data collection systems, high speed communications and small to medium scale forecast modelling. Weather graphics will portray current data and forecasts of local events and high resolution animation. This will be made possible through a new generation of government computers and processors and will include advanced doppler radars and communications at very high baud rates. This new radar system, for example, called Nexrad will generate a tremendous amount of data at every individual radar site. It will show very accurately the height of the showers and thunderstorms, it will produce a graphic of instantaneous surface rainfall rates, surface rainfall accumulation over different periods of time such as, 1 minute, 10 minutes, 1 hour. It will show wind shear areas. In all, there will be 30 products available from every radar site every 5 seconds. And all of these parameters will be available in much higher resolution than anything available now. You will literally be able to see some variations in weather from neighborhood to neighborhood and watch it change from minute to minute with almost continuous mapping.

Not only will this provide a tremendous number of graphics for possible local display on television but some of this data will help properly trained and experienced forecasters in predicting thunderstorms, air pollution, flood events, rainfall and snowfall amounts, rain/snow dividing lines--and much more. Another part of AWIPS-90 involves wind profilers. These new devices will take upper air observations every two minutes compared to every 12 hours now. Because upper air observations are only taken twice daily or 12 hours apart, the main computer models that serve as the basis for today's forecasting activities are only run twice in a 24 hour period. Under this new system, these observations will be taken 720 times in a 24 hour period. This will help measure small scale systems that now slip between observation sites, it will improve forecasts by spotting trends earlier and, of course, will lead to many more interesting graphic presentations for on-air use. Television will literally be able to show a cold front moving across a city and the impact it is having. Graphics will show the variations in rainfall on a minute-by-minute basis as a thunderstorm moves across a region and almost as it moves across your own house. This will be of particular interest in the severe weather situations when tornadoes or damaging storms threaten a particular area.

Private weather companies will be the leaders in bringing this new technology to broadcasters. In fact, the National Weather Service has in the last couple of years increasingly recognized the role of private weather companies. Furthermore, the AWIPS-90 proposal is strongly built around contributions by the private sector and in fact lays out the role for private companies such as ours to be the distributor of data and graphics.

How much is all this going to cost you? Well, weather support is certainly a lot more expensive today than it was 10 years ago but on the other hand, over the last couple of years, the fees and prices for weather support has come down dramatically. For example, when we first began offering our graphics service, we charged $5 per graphic image. Now our fee is $2 per image and for large users who receive graphics through satellite delivery systems, the fee can drop to as little as 20 cents per image. Likewise, we have been able to reduce fees for database connect times. In fact our database connect times have traditionally had a penalty charge built in for overtime use. However, because of the decreasing costs of computers and the increasing sophistication of software we were last month able to completely eliminate our overtime penalty charges. Further, we and the other private weather services have been able to reduce significantly the cost of acquisition of National Weather Service circuits through satellite delivery. As one dramatic example of the kind of decreases in price that have occurred, Siscorp, a company that we represent, now offers a PlainPaper Facsimile, for a mere 12% of traditional Alden Facsimile. This represents a savings of as much as 88%. Also as modems become faster and many stations upgrade from 1200 to 2400 baud and eventually to 9600 baud modems, the cost of data acquisition and graphic acquisition from private weather services will continue to decline. However over the next five years I believe your total cost of weather data will increase as many more types of graphics and data continue to become available, through innovative developments like the Accu-Weather Newrad system because of the debut of AWIPS-90--and because you will probably do more weather programming.

COMPUTER CONTROL PROVIDES RAPID AND ACCURATE GRAPHICS DISPLAYS FOR ELECTIONS AND SPORTS COVERAGE AT THE ABC-TV NETWORK

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ABSTRACT

This paper describes the operation of ABC's Election Graphics Computer System in New York. During Elections and College Football, this facility is used to organize raw vote totals from the News Election Service or game scores from correspondents in the field, and control broadcast graphic devices to create real-time on-air displays.

Through the use of modems, production trucks in the field can also access this information to control graphics devices on location.

INTRODUCTION

Creating graphics displays that contain large amounts of continuously changing information can be both time consuming and error prone. By the time such a display is created manually, the information content is most likely to be outdated. Multiply this condition with many such events going on simultaneously, and you face a major burden. An example where one is faced with these conditions is Elections, where there are hundreds of races with continuously changing vote totals going on at the same time. Reporting live College Football scores on a Saturday afternoon is similar, as there are approximately 40 games that have to be kept track off.

What is needed is a computer that complements the graphics device by keeping track of the changing data, and supplying this data in a form usable to the graphics device. This is the concept behind the Election Computer Facility that has been in operation at ABC for the last ten years. This facility has recently undergone a complete face-lift, bringing it up to date with current technology and expanding its capabilities. In the first section of this paper a technical description of the facility will be presented. Section two will explain the Election vote reporting process, section three the operation of the Election System, section four the College football score displays and section five the Scoreboard Interface computer

FACILITIES DESCRIPTION

In the heart of the Election Computer Facility are two Charles River Data Systems super micro computers running in parallel. The software for the computers was written by Dubner Computer Systems. Although the facility is redundantly equipped to survive failures, loads are shared to speed up normal operations. The system has the ability to keep track of a thousand races. Graphics devices that can currently be supplied information are Dubner CBGs, Chyrons, and Quantel 6030s. Eight CBGs, three Chyrons and four 6030s can be supplied information simultaneously, without degradation of performance. Information can be entered into the system from remote computers through phone lines and modems, or terminals located throughout the plant. Race results can be monitored on video monitors through a special computer called the Nineboard Display System (this computer will be referenced over and over in this paper in different uses. It is a computer built by Walter Bohlin of ABC Engineering, and is used to convert data to video, interface CBGs and Chyrons in the field to Election computers in N.Y. through modems, and interface scoreboards on location to graphics devices). This computer converts data into a nineboard video display, that will display nine races on a single video monitor. Each of these computers can drive eight video monitors, so that seventy two races can be displayed from one computer. The Election Computer Facility uses three of these, enabling two hundred sixteen races to be displayed on the monitors of the Election facility for easy reference at one time. The video from this computer is also distributed throughout the plant in video and RF form so that production personnel can continuously monitor the progress of the elections.

The interface to each graphics device is machine and location specific. As the CBGs and 6030s are used in the plant, direct data connection is possible. Remote Chyrons and CBGs are connected through modems and phone lines, to the Nineboard Display computer. Lets examine each device separately:

CONNECTION TO CBG

The computer system is interfaced through the RS-422 link between its keyboard and mainframe. Every character entered at the CBG keyboard passes through the Election System computers. Under normal conditions the Election computers are passive and pass the characters unmodified to the CBG mainframe. When a special character is pressed on the CBG keyboard the Election System computer interrupts the data flow to the mainframe, and sends the result of the command from the keyboard to the CBG mainframe as KPL commands (The programming language of the CBG). A local data monitor is used to echo the response of the Election System computers.

CONNECTION TO 6030

This device allows multiple control boxes to be connected to its mainframe. The protocol between a control box and the mainframe is RS-422. The Election Computer system emulates a control box by sending the 6030, commands, as if they were send by a control box.

CONNECTION TO REMOTE CHYRON AND CBG

The Graphics devices at ABC's local station WABC-TV and remote trucks fall into this category. In the case of our local station it is possible to run a wire between the two buildings, so a local modem is used to increase reliability and reduce phone charges. Remote trucks dial in over phone lines to access the Election System computers through modems. The Nineboard computer at the remote location is used to convert the data received over the modem into video, for display on local monitors for the production people, and control commands that allow this information to be generated by the attached graphics device. The Election System computers update the remote Nineboard computers whenever a score is changed or every two minutes whichever occurs first. This buffering of the data at the remote Nineboard computer improves reliability as a phone line failure will not cripple computer control of the graphics device, and reduces the processing burden on the Election system computers.

THE ELECTION VOTE REPORTING PROCESS

The basis of the Election vote Reporting process originates from the News Election Service. This organization is a cooperative news agency owned and operated jointly by the news divisions of the three networks, UPI and API. Its function is to collect, tabulate, and distribute unofficial election night vote results as soon as they are available from the earliest source. NES accomplishes this by having a precinct reporter at every one of the 95,000 precinct across the country. The precinct reporter works together with the local election officials and calls in the precinct vote totals to one of three regional NES centers in the country. Votes collected by the computers of each regional center get transmitted to the NES National Center in New York. The computers at the Election Computer facility at ABC are hooked up to the NES National Center computers through data lines. Every five minutes throughout the election night the ABC computers get updated with new vote totals.

The information received from NES is used to create displays that show actual vote totals. These vote totals do not come in until the polls are closed. This depends on the poll location, but can be anywhere from five to nine P.M.. So how is it that ABC News is able to project winners as soon as the pools close? (The information is known actually before the polls close but ABC News, as a policy decision will not project winners until the polls are closed) Besides NES, ABC depends on exit polls to project the outcome of an election. An exit poll is a questionnaire given to a statistical sample of people at statistically determined locations throughout the country after they have voted. The questionnaire contains questions about age, race, sex, religion, income level and the candidates they have voted for. Information gathered from these exit polls, together with statistical data about the candidates and NES vote totals get entered into a simulated model of the election, running on the computers of the ABC Political Unit. As the Election day progresses this model becomes more and more accurate. When the statistical results of a race point out that there is a likely winner, the chief political analyst makes the decision to declare a winner for that race. As the computers of the Political Unit are connected by modems to the Election System computers, this information gets instantly transferred over. Other information that gets transferred from the Political Units computers to the Election System computers are the demographics information obtained from the exit polls.

THE CREATION OF DISPLAYS FOR THE ELECTIONS

DUBNER CBG

The Dubner CBG is the main graphics device the Network uses to create its election graphics displays. Four different types of vote displays and four different types of demographics displays can be used. The vote displays are:

1. Raw vote displays
2. Winning line-up displays
3. Summary displays
4. Maps

Most of the above displays can be created for presidential, senatorial, house and gubernatorial races for any geographical region in the country. The raw vote displays show the type of race, location, party affiliation, the vote each candidate has received, the number of precincts reporting, the percentage of the vote each candidate has received and, if any have been projected by ABC News as winners. The winning line-up displays show which candidates have been projected as winners for a certain type of race. The Summary displays show the total composition of the house and senate and how many seat were lost or gained by a party. Maps show state by state the party or candidate that has won.

The main demographic displays show a demographic category and a bar graphs for each candidate. The bar graph corresponds to the percentage of voters in that demographic group that have voted for that candidate. There are forty different demographic categories. These displays can be generated for any type of race and location.

Although all of the information needed to create a display seem to be coming from the Election Computer system, advance preparation of the CBG is necessary. First the backgrounds have to be created. Then the names of states, candidates, and races have to be stored on disk. The Election System computer is notified of the locations of all this information on the CBG disk. The following is how a typical display gets built: The CBG operator first presses an unused key on the keyboard. This lets the Election System computer know that the CBG operator is issuing it a command. Now the operator enters "/g,ala". This stands for create a raw vote display for the governor of Alabama race. The Election computer searches its database for the Alabama gubernatorial race and sends the CBG mainframe, the location of the background, the location of the names of the candidates, the location of the state name, the votes each candidate has received, the percentage of precinct reporting, the percentage of the vote each candidate has received, and which candidate, if any has been declared a winner, as KPL commands. The CBG then builds this display in about four seconds. When used in a special mode of the CBG called the preview mode(a mode in which you always are one picture ahead of the displayed picture) the info gets displayed in one second.

CHYRON

Chyrons are used for Election Graphics generation at ABC's local station WABC. The Chyrons are interfaced through the Nineboard Display computer and local modems to the Election Computer Facility at the Network. As the local station is more interested in reporting local races, the Election computers are programmed to accept these races from NES. Only raw vote displays are created on the Chyrons. The races the local station will be covering are displayed on the monitors that are interfaced to the Nineboard Display computer at the local control room. Each race is given a number according to its location on the Nineboard displays. When the graphics for one of the races has to be created, a special key-sequence together with the display number is entered at the Chyron keyboard. Within a couple of seconds, the display is created. The display contains the race name, the candidates names, party affiliation, vote totals, percent of the vote, percent of the precincts reporting and the projected winner if any.

QUANTEL 6030

The Quantel is a still store device. When a CBG callup is made, occasionally a picture to enrich the graphics display is desirable(i.e. a picture next to each candidate's vote total). In the 6030 a disk is prepared with all the candidates pictures. In the database of the Election computers the location of the 6030 pictures corresponding to each race is entered. Now when an operator at a CBG keyboard request a vote display, information to build the display is transmitted to the CBG mainframe. At the same time control information is send to the 6030 to display the pictures of the candidates. The display of the CBG is added to the picture of the 6030 through a keyer to produce the final product.

OPERATION OF THE SPORTS SYSTEM

A College football afternoon is quite similar to an election night. There are typically forty or so games going on at the same time. To keep up with the score changes and to create graphics displays, is too tedious to do by hand. So an extension was written to the Election Computer software to keep track of College Football scores. The data from stadiums across the country is gathered by six phone operators that continuously poll the games and get the latest game results. This information is passed on to a keyboard entry operator who enters this into the database of the computer. The production personnel determines which games results will be displayed in the next segment. Games are numbered to allow rapid callup of many games in a row. The CBG operator is given a list of numbers. When the CBG operator enters "/1", the Election computer interprets this as a request to build a display of the first football game in the list. The Election computer searches its database for the latest scores for that game, and transmits it to the CBG mainframe together with the information needed to build the picture. It is possible to display one game result every four seconds.

During a College Football afternoon, ABC also has a remote truck covering a College football game on location. This truck has to have access to the information stored in the election computer in N.Y., so that the program generated in the field can have the latest game scores keyed on to it. Both CBGs and Chyrons, interfaced through the Nineboard computer, and phone lines to the Election computer room in New York, are used. When the local graphics operator enters a special key sequence together with the number of the game he wishes the score from, the latest game scores, together with team names, ranking and period get displayed on the graphics device. This information gets keyed on the display of the game in progress, to create the desired effect.

SCOREBOARDS

One other occasion where the need for computer control of a graphics device is imperative, is when it is necessary to display the content of a scoreboard on the air in broadcast quality video. The scoreboard contains too much information to be typed manually, and the information changes too rapidly. Another problem is that most scoreboards around the country are of different types and use different protocols. This application is perfectly suited for the Nineboard Display computer. The data line between the scoreboard computer and the scoreboard is analyzed in advance to determine the protocol (RS-232,RS-422,current loop, etc.) Then the data is analyzed to determine how it is organized. The Nineboard computer is programmed to interpret these commands. At the event the Nineboard computer is placed between the scoreboard computer and the scoreboard. Now whenever anything gets displayed on the scoreboard, the graphics device connected to the Nineboard computer receives the commands to generate that image. The Nineboard computer can be programmed to calculate information from the data it is receiving from the scoreboard computer, that is not available on the scoreboard.

CONCLUSION

Complex and rapid display generation, necessitated by todays broadcast market is easily accomplished by the use of control computers. ABC engineering has provided its production staff with a very flexible facility and the tools, to accomplish that goal. Both the Election Computer Facility and the Nineboard Display computer can easily be configured for different information sources and graphics devices to allow continued use for years to come.

USING CUSTOM VITS FOR AUTOMATIC TRANSMISSION SYSTEM VIDEO PERFORMANCE ANALYSIS

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ABSTRACT

Vertical Interval Test Signals or VITS have a long history of usefulness as a means of in-service verification and adjustment of transmission systems. Since VITS are subject to the same transmission distortions as the program occupying the active lines of the picture, VITS can provide an accurate real time assessment of a transmission system under dynamic operating conditions. This paper describes recent efforts to make VITS more convenient to use and occupy less transmission time. With lines 10-21 now available for a wide variety of purposes from digital audio to high speed data, at least one line should be reserved for VITS. Advances in digital generators, waveform monitors and automatic video analyzers make this possible.

HISTORY - BACKGROUND

Vertical Interval Test Signals were first used by networks on Telephone Company video circuits more than twenty years ago. VITS were extended to station intercity STL and microwave systems and for a time made an FCC requirement for TV remote control. They are used extensively on satellite transmissions and now likely to be found on any video signal. The original VITS consisted of stair steps, multiburst and window signals which have now been supplemented by literally dozens of variations.

While this paper deals only with NTSC, other transmission standards can and do regularly employ VITS. In NTSC, the first nine lines of each field transmit vertical sync along with the pre and post equalizing pulses used by receivers to create the two interlaced fields. In early receivers with no circuitry to cut off the scanning beam, raising brightness would cause objectionable diagonal retrace lines to appear on the screen until an orderly sweep is re-established. Since this occurs sometime prior to line 20, only black was transmitted on lines 10-20. Since field 2 begins half-way through line 263, the vertical sync occurring at this point in the scanning sequence is also followed by ten lines of blanking.

In all recent receivers the beam is cut off to eliminate this visible retrace, regardless of brightness setting or whether DC restoration is used or not. Vertical sweep circuits generally stabilize scanning by line 16. As a result, it is no longer necessary to transmit black on the lines following vertical sync since retrace is suppressed by receiver design. Video on these lines will appear at the top of the picture when underscanned.

When the FCC established rules for remote control of television transmitters, stations operating by remote control were required to transmit FCC specified color bars, multiburst and composite VITS on lines 17 and 18. The Commission also established that a Vertical Interval Reference (VIR) signal is the only signal that can be transmitted on both fields of line 19 and later captioning for line 21.

With deregulation, line 19 and 21 continue to be exclusively used for VIR and captioning, however, line 17 and 18 VITS are no longer required for remote control. At present, lines 10-18 and line 20 are available for virtually any purpose with some limitation on amplitude when data is placed on early lines. Transmitting data is particularly attractive since each line can support 9600 baud with substantial overhead and data can be revenue producing. No one has offered to pay stations to carry VITS. With the potential of revenue from transmitting data in the vertical blanking interval and without an FCC VITS requirement many broadcasters are now faced with a problem of retaining any VBI space for test signal use. In fact, some station managers have sold more lines than exist in the vertical blanking interval, not understanding that just like real estate, they aren't making any more VBI lines. This represents a significant shift from the time when engineers had lines 17, 18 and 20 available exclusively for VITS.

VERTICAL INTERVAL TEST ALTERNATIVES

Over the years many different test signals have been devised since VITS can be used for virtually all testing except those requiring field rate signals. As a result, we now find signals for linearity, response, delay, axis shift, quadrature error, tilt, noise, transient response, color phase, and levels of all kinds to name the most common varieties. Assuming that your station does not have the need or inclination to put different full time test signals on each of the 20 available lines, there are two alternatives. Sequencing VITS on one or more lines or the use of combination signals.

SEQUENCING

Presuming that at least one or more television fields is available for VITS use, sequencing or switching test signals either for a specific test or on some set schedule is simple and inexpensive. The necessary test signal can be inserted in the VBI and left in place until another is more appropriate. A good example would be a typical television studio with a transmitter controlled from the studio. With both ends of the VITS transmission system at one location switching tests as needed works quite well.

When the transmission system has multiple destinations, such as a satellite transponder feeding a network, access to several desirable test signals can be provided by switching signals such as modulated steps, multiburst, color bars and window in some sequence, leaving each up for several minutes. Just as with an audio step tone generator, the duration is likely to be found either too long or too short. Sequencing test signals guarantees that the test signal needed will not be available most of the time.

With some newer generators, such as the Tektronix 1910, automatic sequencing can be programmed at almost any rate up to and including color field rate. This can be very useful since one VBI line can accommodate four different test signals with repetition every fifteen hertz. Providing virtually continuous access to each test signal Dynamic measurements or adjustments are possible and display on typical waveform monitors with line select is quite acceptable. A monitor such as the Tektronix 1480 can display each of the four VITS signals individually. Unfortunately while the 1910 generator can put test signals on specific color fields, the 1480 waveform monitor cannot positively select color fields for display and as a result there is only one chance in two of selecting the desired VIT on the first try. On a waveform monitor that can only display both color fields superimposed, the VITS can still be adequately interpreted.

Up to this point, only the use of VITS with conventional waveform monitors has been considered. Automated analyzers such as the Tektronix model 1980-Answer can be used with sequenced VITS, however, color field selection is not positive. This is even more awkward than using a 1480 waveform monitor where at least the desired test signal can be selected with an additional try or two of the line select. Another complication is that some measurements with the Answer system are integrated over many fields. As a result, color field sequencing does not work with Answer. Use of VITS programmed for a duration of several minutes each is satisfactory but would require the operator of the automated system to initiate measurements early enough once the desired signal comes up to allow all sampling to take place before it disappears.

If there is no interest in dynamic testing or if the VITS are not to be used for in-circuit adjustments, theoretically a test signal could be transmitted at widely spaced intervals and used at the operators' convenience with a storage scope or device that could faithfully capture and display the signal. While an abundance of storage devices exist, these are rarely used for television waveform analysis.

COMBINATION SIGNALS

The second and in my opinion the most fertile ground for development of efficient test use of the VBI comes from the use of combined test signals. A very familiar example is the FCC composite which was orignally defined by Section 73.699 of the rules for remote control. This combines several familiar signals of interest in transmission systems. Starting with six modulated stair steps for differential phase, differential gain and low frequency linearity measurements; next comes a 2T pulse for transients and a modulated 12½T pulse for chroma gain and delay which is followed by a window for low frequency gain, tilt and a white gain reference. It should be noted that the window or white flag is a convenient location for source identification. More on this later.

NTC-7 combination (which is similar to the FCC composite but with the position of the window and the modulated steps interchanged) and NTC-7 composite which combines reduced amplitude multiburst and modulated pedestal on one line are also very useful and efficient signals. The original FCC multiburst and FCC color bars, each provide unique and valuable information but occupy a complete line each.

Faced with a shortage of VBI space, at Nebraska ETV, despite FCC deregulation we wanted to have VITS with all the measurement capability of the three FCC defined color bars, multiburst and composite. The custom capabilies available in the Tektronix 1910 digital test signal generator was a way to put the multiple signals needed into no more than two custom signals which could share alternate fields of one VBI line. In the process a relative newcomer, multipulse was compared with the venerable multiburst and found to provide virtually all of the advantages and capabilities of multiburst and significant additional information as well.

While multiburst can provide information on frequency response and axis shift, due to high energy content it has been traditional to operate multiburst at reduced pulse amplitudes to avoid overload in transmission circuits with equalization. Multipulse is actually an extention of the familiar 2T and 12½T pulses which are joined by sine squared envelopes modulated with several other frequencies of interest. For transmission the best multipulse frequencies are 1.25 MHz requiring a double width or 25T envelope and 12½T pulses modulated with 2, 3, 3.58 and 4.1 MHz. This signal provides very good frequency response information and can provide quantitative group delay information for frequencies in addition to the color subcarrier.

It also happens that the multipulse high frequency energy content permits transmission at full level. The pulse shapes are fixed but by reducing the blanking space between pulses, it is quite possible to reduce the overall duration of the multipulse to a little more than half a line. This was found to leave sufficient room for a horizontally sequenced version of the FCC color bars including its white reference flag. This combined signal which we call Multipulse/Color Bars, (Figure 1) could be used as a companion to the standard FCC composite. It should be noted however that since the FCC composite includes a 2T and 12½T pulse which duplicate part of the multipulse signal, there would be unnecessary redundancy between these two signals. Therefore, the FCC composite has also been modified replacing the the 2T and 12½T with a reduced duration modulated pedestal which can be fit into exactly the same time period. This is called MP Modified FCC Composite,(Figure 2) with MP referring to Modulated Pedestal. Modulated pedestal is used for measuring chrominance to luminance intermodulation and for chrominance non-linearity gain / phase distortions which are common in envelope detectors. Adding the modulated pedestal compensates for the axis shift measurement capability lost when multiburst was replaced by the otherwise more flexible multipulse. Satisfied with the definition of these two custom test signals, Multipulse/Color Bars and MP Modified FCC Composite, detailed information was provided to Tektronix recommending as little change as possible to standard rise times, modulating frequencies, phases and amplitudes so that users would find the signals to be familiar. Each signal includes its own white reference, which is helpful in measurements with conventional waveform monitors. It should be noted that the lowest multipulse frequency is 1.25 MHz which is useful in comparing vestigal sideband amplitudes and that the highest multipulse frequency is 4.10 MHz which is more useful for both amplitude and group delay measurements in systems with rapid roll-off characteristics above 4.2 MHz.

In the Tektronix 1910 generator, programming for all test signals is contained in five E-Proms. The reprogrammed proms obtained from Tektronix replace two little used test signals with the multipulse/color bars and MP modified FCC composite signals just described. All other signals remain available and unchanged.

We have operated our nine station Nebraska ETV Network several months with these custom signals available in addition to the "standard" FCC composite full duration multiburst and color bars. This has allowed our transmitter personnel to compare and critique these custom VITS. The results have been favorable and for our needs no further changes were found necessary.

The next and by far the most time consuming step was to develop the software changes in the Tektronix Answer so that automated measurements could be made using these two custom signals. This is required since three of our transmitters incorporate Answer as the only means of visual modulation monitoring. Their control points do not have off-air monitoring. This software was written in our shop with the cooperation and assistance of the manufacturer. This system has been completely implemented and meets our requirements very well. VITS, necessitated by our transmitter remote control and monitoring system, have been confined to only one VBI line, line number 20. It not only retained all of the original capabilities which were available when they occupied line 17 and 18 but have been enhanced by the advantages of multipulse and the addition of modulated pedestal.

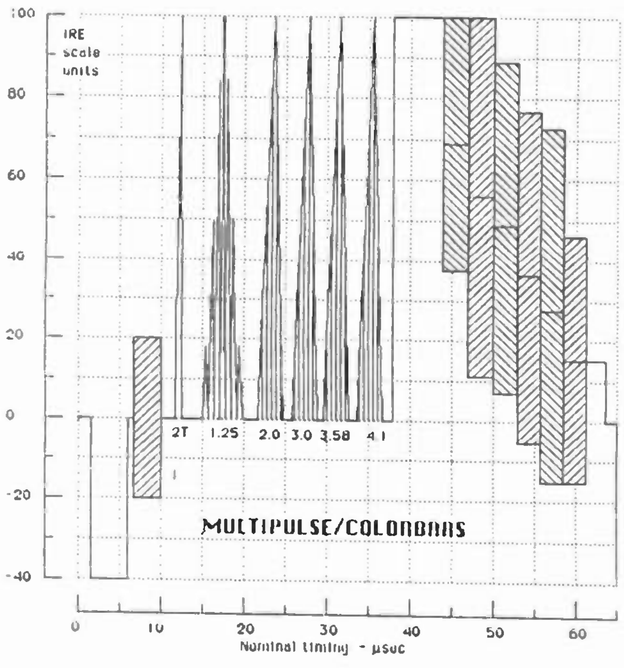

Figure 1

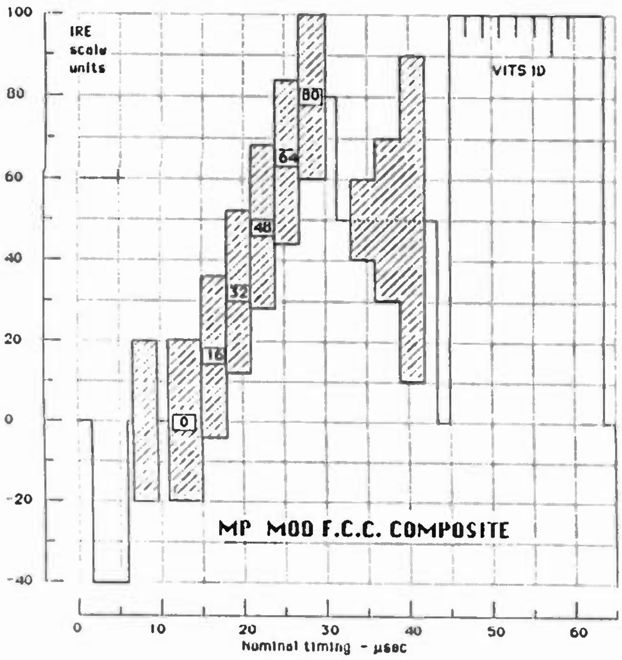

Figure 2

SOURCE ID AND AMOL

Considering the difficulty of consolidating VITS, it is certainly our intent that the remaining VBI space be used very wisely. While neither VIR-Vertical Interval Reference on line 19 nor captions on line 21 are using their allocated space as efficiently as we would like, both are valuable services to the public and any change that would adversely affect their use should avoided.

Automated Measurement Of Line-Ups or AMOL requires only very slow data transmission rates to provide the date, time and origination point information needed for this internal broadcasting industry application. Source identification similarly needs to convey very little information and from my prespective should occupy little VBI space.

The PBS Satellite Distibution System uses a very effective source ID system consisting of two levels of inverted pulses at the top of the VIT window. Originally these were generated in a Tektronix 149 test signal generator which was equipped with a source ID option. Dip switches selected the appropriate code. Since the 149 is no longer available, Nebraska ETV again had the Tektronix 1910 proms reprogrammed to generate this source ID. While not as easy to change as dip switches, a source ID generator will likely live out its life at one location anyway. About ten PBS locations are currently using 1910's with this source ID. The software to generate these custom test signals with the 1910 is available at nominal cost from Tektronix. Similarly users of the 04 software version Answer systems can obtain the 08 update software from Tektronix for automated measurement of the two test signals.

CONCLUSIONS

While our work was done with Tektronix equipment, it should be noted that most digital test signal generators built by other manufacturers can also be customized. With Magni, the multipulse/color bar and MP modified FCC composite have been duplicated on the exhibition floor. North Carolina ETV uses Marconi automatic signal analyzers operating on these two Tektronix generated VITS.

Vertical Interval Test Signals are alive and well. They have acquired a greatly enhanced ability through use of digital technology and can be tailored for specific needs, both now and for those yet to be defined.

ACKNOWLEDGEMENTS

This author wants to express his appreciation for the assistance provided by Tektronix in developing the software for the generation and automatic anaylsis of custom test signals described. Specifically John Kelley and Larry Harrington of Tektronix have been very helpful in providing equipment and technical assistance, and Gary Praeuner of the NETV Maintenance staff for his work in modifying the Answer 04 software for the custom VITS and Carol Lederer for her perserverance in preparing my manuscript.

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NEW DEVELOPMENTS IN A COMPATIBLE HIGH DEFINITION TELEVISION TRANSMISSION SYSTEM

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ABSTRACT

A compatible HDTV transmission system is under development at NYIT. This paper describes the resolution that can be achieved, the signal format, how compatibility of aspect ratio can be accomplished, and how the display can be provided with progressive scan.

Introduction

In previous years a compatible HDTV transmission system has been described.[1,2] This system has been designed based on exploiting data gathered from extensive psychophysical measurements of vision. The transmission consists of a standard 525-line color signal, that we will call "base-band", plus a second channel that contains detail information. This second "detail" signal contains all of the information necessary to upgrade a standard 525-line image to full resolution HDTV. The second signal is relatively narrow- band, since it is transmitted at low frame rates and uses diagonal sampling. This system lends itself to the use of a progressively scanned detail camera tube and can be easily displayed with progressive scan. Under these conditions the sharpness of the image is higher than HDTV images derived from interlaced cameras.

This paper addresses a number of questions asked about the system - namely:

1. What is the transmission spectrum?
2. How does one prevent registration problems in the image caused by propagation delay differences between the base transmission and detail signal?
3. How is it possible to make the 3x4 aspect ratio of NTSC compatible with the 3x5 aspect ratio of HDTV?
4. How can progressive scan in the camera and in the display be used without producing motion artifacts?
5. What is the resolution achievable with this system?

Transmission Format

The base-band channel of the transmission is a 525-line color signal. This would be a standard NTSC signal for terrestrial broadcast and cable transmission. For satellite transmission, it could be in NTSC format or, alternatively, in analog component format in order to yield a better signal-to-noise ratio. The additional detail transmission is in analog component format in all cases. Many two-channel bandwidth systems require that the channels be adjacent and that the entire spectrum be transmitted from a single antenna location. This is not a restriction for the two channels we propose to use. In many cases the detail transmission can be adjacent in the spectrum to the base-band signal. This would probably be the case for satellite, pre-recorded media, and for terrestrial transmission where an adjacent channel is available. When an adjacent channel is used, the detail transmission can use the carrier, synch and color burst of the base channel.

For terrestrial transmission and in cable transmission, it may not be possible to locate the detail transmission adjacent in frequency to the base channel. It may also be desirable to have the detail antenna at another location than the base channel antenna. For example, a VHF station may have its detail transmission in the UHF band. This is shown in the block diagram of Figure 1. In this case a code in vertical blanking is transmitted on the base channel that tells the HDTV receiver where to tune to find the corresponding detail transmission. If no detail is received (i.e., if the UHF signal is too weak or if the station does not have a detail transmission), the HDTV receiver will simply display the NTSC signal scan-converted to 1125 lines, but

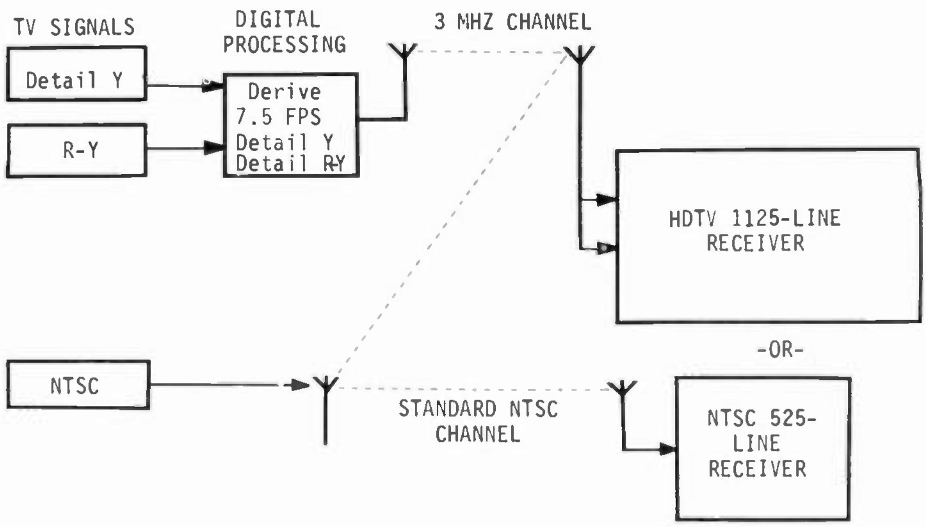

Figure 1. Proposed HDTV Compatible Transmission System

at standard resolution.

In the case of a non-adjacent detail transmission, the second signal has its own carrier, frame index and burst. Therefore transmission path delays for detail need not correspond to those of the base signal. The detail is clocked into a store in the receiver, based on its frame index and burst. It is clocked out, based on the synch and burst of the base-band transmission. Care must be taken to design the frame index code and burst signals so that they are at a lower amplitude than the video signal. This will minimize adjacent-channel and co-channel interference.

The detail signal is diagonally sampled as shown in Figure 2. The sampling is done at a clock rate of 3 x 3.58 MHz and transmitted with a bandwidth of 5.3 MHz and a frame rate of 15 fPS progressively scanned. If further bandwidth compression is needed a frame rate of 7.5 fPS can be used for detail. If 7.5 fPS is used, two stations can time-share a channel. Alternatively, half the clock frequency can be used with a detail bandwidth of 2.7 MHz.

The resolution of the NTSC, B-Y color signal is adequate for the B-Y signal in HDTV. However, the R-Y detail signal in NTSC has inadequate resolution for HDTV. Therefore we propose to transmit an additional R-Y detail signal along with the detail luminance transmission. This R-Y detail signal is derived from the R, G, B camera.

Compatibility of Aspect Ratio

Although 525-line images have a 3x4 aspect ratio, and 1125-line HDTV images have a 3x5 aspect ratio, compatibility between these two formats is possible. Figure 2 depicts a method for achieving aspect ratio compatibility. The 525-line image allows video to invade the horizontal blanking interval so that active video lasts for 56 microseconds instead of 52 microseconds. It starts immediately after burst and ends immediately before H synch. This provides an additional 8% more horizontal width. This part of the signal will be blanked by the 525-line receiver, but displayed by the 1125-line receiver. Since the HDTV receiver must scan convert the 525-line signal to 1125-line format, its time-base correction can make H blanking in the 1125-line display any interval needed by the picture tube.

In scan converting the 525-line image to 1125 lines, line interpolation is used. The low resolution portion of the HDTV image is obtained by scan converting the 525 line image to 1125 line format. This process uses 440 active lines of the 525-line format with a scan conversion ratio of 2.33 to 1. This provides 1024 active lines in the HDTV display, derived from 88% of the lines in the NTSC signal. This "crops" the height of the 525-line image by 12%. The combination of the 8% increased width and 12% decreased height provides the HDTV receiver with a 3x5 aspect ratio. This process has the advantage that only 8% of the width and 12% of the height need be protected when

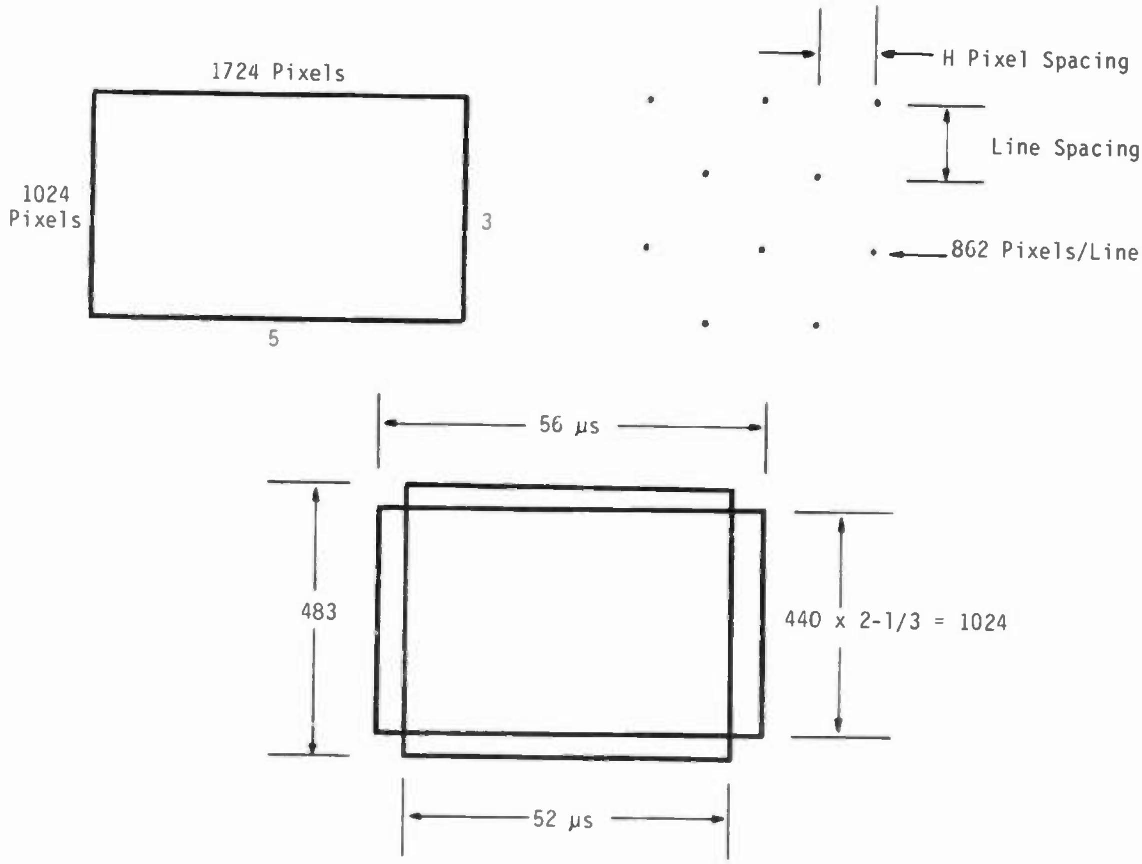

Figure 2. (Top Left) HDTV Detail Signal Pixel Format.
(Top Right) Magnified View of Pixel Format.
(Bottom) Method for Achieving Aspect Ratio Compatibility for 3x4 and 3x5 Video Signals [See Text for Explanation]

shooting program material to be displayed in either format.

Camera with progressively scanned detail

It has long been recognized that camera tubes provide much higher vertical resolution if they are progressively scanned. The signal processing techniques used by this system make it possible to produce a color HDTV camera that uses progressive scanning for the detail signal. Because of the slower scan rates for the detail sensor, the bandwidth and sensitivity of this type of camera can be improved over that of a standard interlace-scanned HDTV camera.

An HDTV camera has been constructed as shown in Figure 3, and was demonstrated at the 1986 NAB convention. It uses a standard 525-line R, G, B camera with a fourth tube progressively scanned at 15 fields/sec (fPS) in order to derive detail information. The high-definition camera signal is spatially filtered with a 2-dimensional digital matrix to provide the detail image. The temporal response produced by integration on the face of the camera tube is adequate for 15 fPS detail signals. However, if a 7.5 fPS detail signal is used, it is temporally filtered using frame stores from the 15 fPS signal. Using the lower frame rates necessitates "shaping" the temporal response by oversampling and pre-filtering temporally, to avoid visible temporal aliasing of detail information.

HDTV Receiver

In the receiver a monitor with 1125 lines/60 fields inter-laced can be used. Alternatively, the NYIT system can easily provide signals to be used in a progressively scanned display with 1125 lines/60 frames per second.

Compared with the interlace display, a progressive display eliminates interline

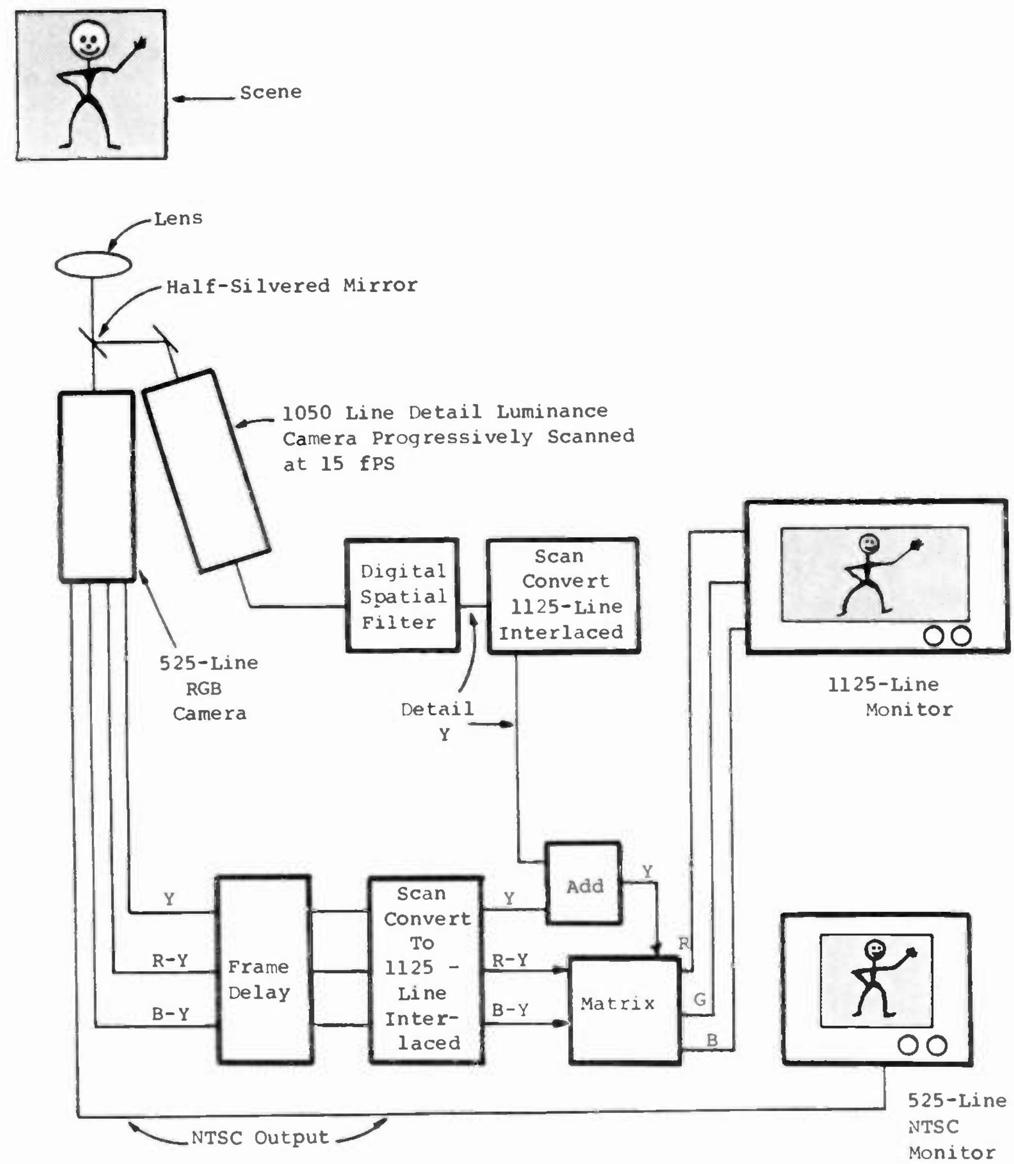

Figure 3. NYIT's Dual Channel HDTV Camera with its Digital Signal Processing.

flicker, and reduces the visibility of the field-line structure that is noticeable when a viewer makes vertical eye movements.

Other HDTV systems using progressive displays have found it necessary to use motion-adaptive processing in the receiver. This avoids the intolerable motion artifacts that stem from using information derived from the frame to produce an 1125-line field. If only the current field information is used, the system has good motion rendition, but suffers a loss in vertical resolution.

If, however, low resolution information is derived entirely from the current field, and the remaining detail information is derived from the stored detail signal, the display can then have progressive scanning at full resolution <u>without</u> motion artifacts.

A block diagram of the HDTV receiver with an interlaced display is shown in Figure 4. A signal is produced that drives a standard 1125-line HDTV interlaced monitor. The 525-line signal is scan-converted to 1125-lines interlaced format using line interpolation. All of the

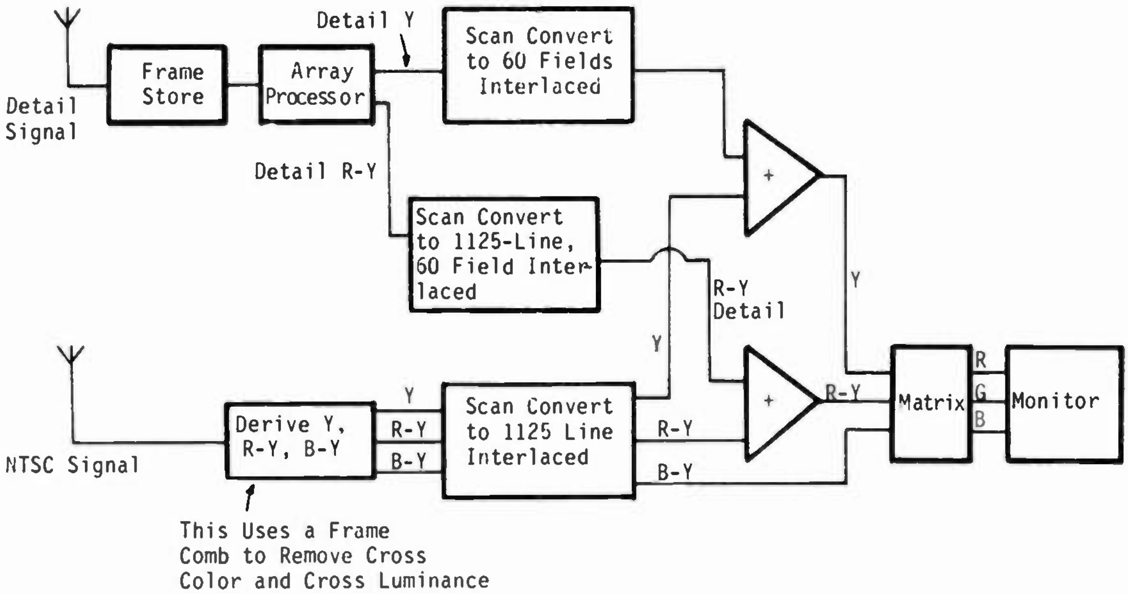

Figure 4. Compatible HDTV Receiver Signal Processing with Interlaced Display.

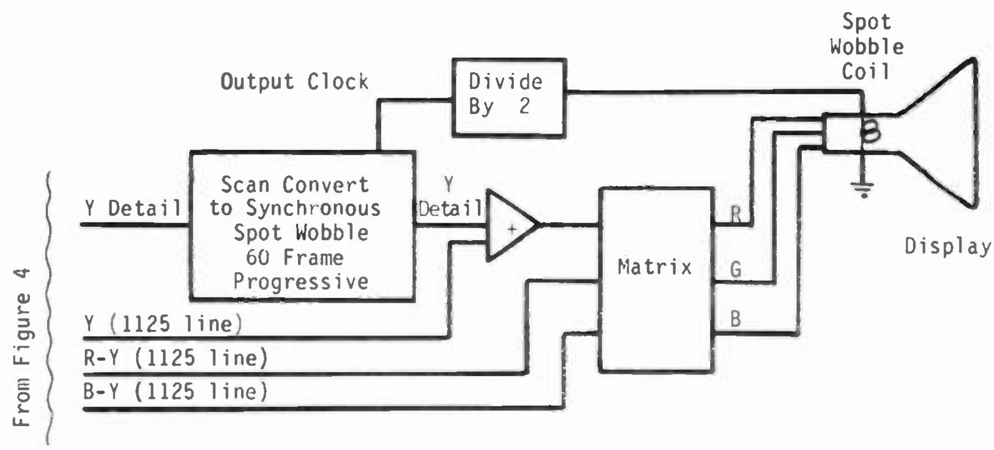

Figure 5. Progressive Display Option for Compatible HDTV Receiver (Signals for This Block Diagram Derived from Figure 4)

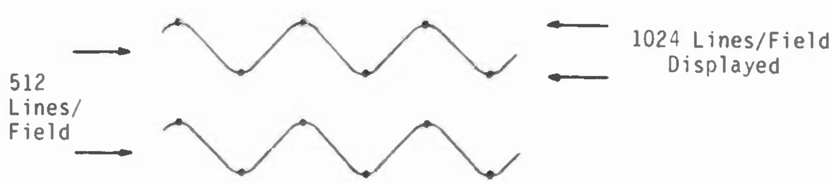

Figure 6. Progressive Display Pattern Using Synchronous Spot Wobble to Display Two Lines Per Horizontal Scan.

displayed low-resolution information is derived from the current field of the 525-line signal. The stored 1125-line detail signal is scan converted to 1125-lines interlaced, and simply added to the other signal for display.

In Figure 5 a block diagram of a system using progressive scan is provided. A standard 1125-line interlaced monitor is still used at the same scan rates as above. However, it is equipped with a small deflection coil that will wobble the spot vertically by one line-width at a frequency of half the output clock of the detail store. Because of the diagonal sampling, two lines can be clocked out during one scan as shown in Figure 6. The 525-line signal is scan-converted to this new format as before. However, two of the detail lines are clocked out of the detail store in alternate dot sequence. This provides 1024 active scan lines each field, and eliminates interline flicker and line crawl that occurs with interlaced displays. It also provides good motion rendition and high resolution, without the need for adaptive processing in the receiver.

Progressive scan can also be provided without spot wobble, by scan-converting both the detail and the current field of 525 line information to 1125 line/60 frame progressive format and adding them together.

Achievable Resolution

The NYIT compatible HDTV transmission system can derive its signals from either an 1125-line interlaced camera, or from a recording, and can be displayed on an 1125-line interlaced monitor. Scan-conversion from these inputs can be done by replacing the storage on the face of the detail camera tube with electrical frame stores. Interlaced scanning of the camera and display, however, result in limiting resolution of about 700 TV lines, both vertically and horizontally.

Using a progressively scanned detail camera combined with a progressively scanned display as described above, the limiting resolution that can be achieved with the pixel format shown in Figure 2 is 920 lines, both vertically and horizontally.

Conclusion

In this paper many of the questions that arise in the design of a compatible HDTV transmission system have been addressed. The system has been designed in such a way that it is not only compatible with NTSC transmission, but also compatible with 1125-line source material and displays. Compatibility of the 3x4 and 3x5 aspect ratios is possible. The system has also been configured to take advantage of the improvements that can be achieved using progressively scanned cameras and displays.

While much work remains in order to completely implement this compatible HDTV transmission system, most of the critical questions have now been resolved.

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IMPROVED ON-AIR NTSC THROUGH COMPATIBLE CHANGES IN STUDIO STANDARDS AND PRACTICES

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INTRODUCTION

The technical specifications for the NTSC color television system were established in the early fifties, and at that time the parameters given were precise enough to satisfy the tube technology existant at that time. In fact it was quite obvious to engineers working with early NTSC hardware that adhering to the recommended practices, and the tolerances given, were almost unachievable without constant monitoring and frequent readjustment. Vacuum tube circuitry in the fifties and sixties were neither adequately stringent nor stable enough to meet the theoretical limits of NTSC capabilities. Over the three decades that NTSC has been in operational use, the component technology that can be applied to it has dramatically improved. Compact, stable solid-state circuitry has removed the problem of staying within stringent signal tolerances for long periods of time. Nevertheless, these improvements in technology, which now permit NTSC to operate close to its ideal imaging capabilities, have not led to any new examination of the original signal specifications, to see where more stringent definitions could be imposed to improve overall operations and thus enhance the end user's picture quality.

This paper presents a three-way approach that can be used to significantly improve the characteristics of the NTSC color system, while maintaining full compatibility with the current RS170A specifications as defined by both the FCC and the CCIR. The results aimed for are to provide a cost effective means by which hundred of millions of home viewers can see incrementally better NTSC images, no matter what kind of color television receiver they already own.

This three-layer proposal includes suggestions that: (a) the existing NTSC rules should be better observed in all phases of signal processing; (b) that certain implicit rules that are only implied by the nature of NTSC need adherence; and (c) some compatible beneficial changes can be made to the rules to improve picture quality.

OBSERVANCE OF SPECIFIED NTSC RULES

There are two general areas where observance of NTSC rules usually apply in varying degrees.

At the transmission level adherence to rules is almost automatic because of the built-in limitations of the equipment, and the need to meet proof of performance requirements of the FCC. Little complex signal processing is done at the transmitter when the NTSC signal comes in over the STL (Studio Transmitter to Link). As a result the quality of the radiated NTSC signal may be a function of the few elements that modify it at that site, the final encoder and the hopefully precision sideband filters that shape the selected channels response curve.

However, the NTSC signal path through the studio is a much more convoluted one. Here the signal is routed, switched, recorded, and processed through successive devices in both its analog and often digital form.

Within the studio or production facility, there may not be any regulations for common practices to maintain the NTSC baseband video at its optimum, and each of the devices used to manipulate the signal may indeed contribute some unwanted deterioration to the overall system.

As an example of some studio devices which contribute to this growing problem, consider the proliferation of character generators, computer-graphic systems, color keyers and digital effects systems. Most of these systems, which have entered into the mainstream of program production, use such fast rise time video signal edges that they generate illegal sidebands when they are applied to the chroma channels of an encoder.

The I & Q (or R-Y and B-Y) chroma bandwidth bounds are exceeded, and the resultant signals, full of intermodulation overlap, can never be properly decoded by even the best comb filter decoder in a monitor or receiver. Obviously the careful control of these factors at the source would contribute greatly to the betterment of the NSTC images received by the home viewer.

EXPRESSION OF IMPLIED RULES

Throughout the complex studio chain, the NTSC signal finds itself submitted to a variety of sampling mechanisms which are subject to the Nyquist criteria. The techniques for minimizing these effects have been described by researchers in America, Japan and West Germany. In fact recent articles in the SMPTE Journal, by Dr. Wendlund and his colleagues, and in the IEEE Communications Journal, by Takahiko Fukinuki, et al., give good theoretical analyses of these phenomena.

The answer to the problem lies in the use of very careful multi-dimensional Nyquist pre- and post-filtering. This would result in the elimination of aliasing caused by line scanning, and by the 2:1 interlace in normal television. It would also reduce the motion artifacts and the chroma/luminance spectral overlap that creates visual disturbances in the image when certain kinds of fine details are present.

When NTSC was first developed the cameras of that era were incapable of rendering any useful MTF at luminance frequencies above 4.0 MHz, therefore, the notch filter used in the home receiver to filter out the color subcarrier did little harm to the overall resolution of the image. That is no longer the case, and modern cameras do have useful high frequency output above 6.0 Mhz, therefore, requiring that the upper luminance frequencies be recovered after comb filtering of the chroma signals. Again, this implies adherence to operational practices that retain the full quality of a proper NTSC signal.

COMPATIBLE RULE CHANGES

It is also possible to make some beneficial changes to the current NTSC rules which still maintain full compatibility with the present system. To make this clear, "compatible" in this case means full forward and reverse compatibility with no degradation if any new element is introduced in the chain, either at the transmitter or the receiver. The level of any improvement will increase if both sides are implemented.

It has already been adequately demonstrated that the use of 2H comb filtering in the encoder and decoder, in combination with some additional adaptive logic circuitry, can indeed produce a near RGB result in the home receiver. That technique has by no means been taken to its limit, and laboratory tests have shown that further improvements can be made with more than 2H combs. So far, the 2H comb is a cost effective means of getting a considerable improvement in NTSC images without burdening the system with greater complexity and cost.

Another proposal is to replace the currently differing I and Q bandwidths with equal 1.0 MHz baseband channels, which would also have very sharp roll off characteristics incorporated into them. This would definitely help to eliminate the effect of cross luminance in the decoder. If at the same time the transitional characteristics of the chroma channel were very strictly defined, there would be a significant reduction in those well known deficiencies of NTSC, chroma ringing, chroma/luminance delay and the fast rise time pulse handling. These high quality NTSC signals, radiated by a transmitter that has good filters, will certainly make a visible improvement even on old notch filter receivers.

Speaking of transmission, that may very well be the weak link in the chain, and we would propose very strict performance regulations for filters used in the video channel of the transmitter. This would assure that what is actually transmitted on the air would not negate all of the good practices adopted in the studio to improve NTSC image quality.

SUMMARY

In summary, it is clearly evident that the implementation of these steps for the originated and trasmitted NTSC signal would greatly improve the quality of the signal, and would greatly simplify the design of more efficient home receivers.

Line doubling at the receiver is made easier and better if the transmitted signal has been properly pre-filtered in the scanning process; sophisticated comb filter decoding, making use of adaptivity in the horizontal and vertical domains, and of chroma bandwidth expansion, is made easier if chroma luminance spectral overlap is not allowed in the transmitter.

In conclusion, it is believed that an NTSC signal, fully compatible with the present standard can be displayed as a 525 line, 60 Hz progressive scan image. This image will be free of ringing, cross color, cross luminance, and have all appearance of a 7 MHz RGB signal.

These changes in quality, reasonably easy to implement, will give to NTSC a longer lease on life as it was previously expected, and will give to this industry enough time to design, without unnecessary haste, the proper high definition television system of the future.

National Association of
NAB
BROADCASTERS

UHF SUPER POWER AND THE KLYSTRODE— THE BROADCASTER'S COMPETITIVE ADVANTAGE

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The over the air broadcaster today faces serious challenges from other forms of video delivery. This paper addresses a weakness in current UHF transmission systems and proposes a technique for improving the broadcasters competitive position. A detailed discussion is presented which compares high gain versus low gain transmitting antennas. The comparison is made for a typical 5 megawatt ERP station.

A 10db to 20db advantage in received signal strength for low gain transmitting antennas is developed. The need for significantly high transmitter output power is also justified.

High transmitter power demands the energy efficiency of the klystrode and the reliability of waveguide transmission and antenna systems.

The measured performance of a new klystrode powered transmitter is also presented.

I. INTRODUCTION

The UHF broadcaster today is faced with unprecedented market challenges which threaten to take significant portions of his audience away from traditional "over the air" program delivery viewing. These new technological competitors for viewers' time come in the form of low cost video tape and cable, or satellite delivery systems.

While sophisticated advances in broadcast technology such as HDTV are under development, immediate help may well be found by correcting one of the weaknesses of the basic TV transmitting plant employed by the majority of stations currently on the air.

This paper will discuss the traditional 5 megawatt ERP station from the standpoint of ease of reception and coverage. It will investigate current antenna designs and their coverage limitations.

Finally, this paper will propose the concept of super power using the new Klystrode amplifying device in combination with highly reliable waveguide, low gain, antennas to produce high field strengths over large geographic areas. The advantage of this super power approach is to permit low technology receiving installations to produce high quality pictures with a minimum of viewer interaction.

The hardware necessary to produce and radiate super power signals will be described with special emphasis on the klystrode vs klystron trade off. Details of a new klystrode transmitter will also be presented.

II. THE STANDARD UHF TRANSMITTING PLANT

The conventional UHF TV transmitting plant in use today generally employs klystrons to produce 60kW to 120kW of RF output power and a high gain antenna to produce a focused field up to the FCC limit of 5 megawatts ERP. Unfortunately, the highest radiated power occurs only in the center of the narrow vertical beam of the antenna. Today's standard antenna design philosophy assumes that the center of the vertical beam is aimed at the radio horizon. (Some 40 miles away for a 1000 foot AGL antenna) Null fill techniques are used along with electrical or mechanical beam tilt to provide a uniform field distribution from the radio horizon back to close into the antenna base. Thus, a theoretically uniform illumination of the coverage area is achieved.

In some cases, the center of the vertical beam is aimed inside of the radio horizon to provide maximum illumination of specific population areas, sacrificing grade B coverage.

It is well known that the higher the horizontal gain of an antenna, the more narrow the vertical beam will become, assuming horizontal shape characteristics are to be maintained unchanged.

Thus, a typical antenna with a power gain of 60 may have a vertical beam of approximately 2 degrees. While an antenna having a power gain of 16 will have a vertical beam width of about 7 degrees.

If a low gain antenna is chosen having a wide vertical beam width, significant advantages can be achieved in the station's primary coverage area.

As stated earlier, the use of beam tilt and null fill in a high gain antenna provides somewhat uniform illumination of the coverage area. If transmitter output power is increased to compensate for a lower gain wider vertical beam antenna, the same field strength can be achieved at the radio horizon as with a high gain antenna.

However, as the surface field is examined between the radio horizon and the base of the antenna it will be found that the low gain antenna provides significantly higher relative field strengths. In short, the low gain antenna's relative field is not constant but increases significantly from the horizon to the base of the antenna. Thus, very high fields can be established in the primary coverage area, close in to the station, without sacrificing the grade B coverage.

Figure 1 is a comparison of field strength in dbu versus distance from the antenna for a high gain versus a lower gain antenna system providing the same ERP at the horizon. Both antennas have the same beam tilt.

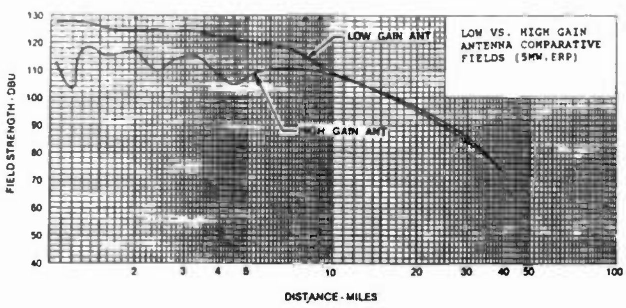

Figure 1 - Field Strength vs. Distance Based on FCC (50,50) Curves

At ranges greater than 10 miles the field from both antennas are approximately the same. Inside of 10 miles the field produced by the low gain antenna is significantly stronger. In fact, the low gain antenna field inside of 10 miles is between 10db and 20db stronger. This is truly a significant advantage.

The higher field intensity, from the low gain antenna, in the station's city grade coverage area, will provide the necessary received signal levels to make program reception possible by even the worst receiving antenna configurations.

Ease of UHF reception, without antenna adjustment is the expected result from the use of super power. This ease of reception is necessary to capture a larger portion of viewer attention and thus viewer time.

III. SUPER POWER HARDWARE

A. The Antenna

It has been shown that lowering antenna horizontal gain will produce a wider vertical beam and thus increase city grade saturation by 10db to 20db. Unfortunately, this also means that more RF power must be handled by the antenna to meet the desired 5 megawatt ERP level at the horizon.

A 60kW transmitter requires an antenna gain of at least 83 to meet 5000kW ERP, ignoring transmission line losses.

If an antenna gain of 16 is chosen to achieve a vertical beam width of 6.8 degrees the transmitter power must be at least 312.5kW. This is significantly higher than the maximum levels of 240kW found in service today.

The power handling capability of splitter fed, slot or dipole antennas, would be severely stressed at 300kW levels. Antenna reliability demands a structure that has significantly higher potential power handling capacity than its expected working level.

Fortunately, recent antenna innovations have produced several designs based on waveguide structures. These antennas rely on the waveguide itself to divide and distribute the input RF energy. Power levels in excess of 300kW are well within their capability.

B. The RF System

Conventional transmitters operating at 120kW and above have utilized waveguide components for both filtering, switching and energy transmission for many years. The waveguide approach permits significant power to be processed without risk of burn out and at high power transmission efficiencies.

The underlying reason for waveguide's efficiency and high power capability is its relatively high impedance. Its ability to carry large amounts of energy for long distances follows the high voltage/low current philosophy used by major utilities.

The use of coax on the other hand, has severe power limitations, as well as specific frequency defined deficiencies. Nine inch coax is limited to 180kW of visual system power* at Channel 39, and 8" to 135kW of visual system power* at Channel 51. When waveguide is examined,

* (Visual system power rating includes a factor of 10% Aural.

the lowest power limitation is for WR-1150, with a rating in excess of 500kW visual system power*. From this fact it becomes obvious that waveguide components, inclusive of phasing/switching systems are mandatory for super power applications. The use of "Magic Tee Phasers", "Variable Ratio Combiners", and other high power waveguide components enable extreme flexibility and redundancy in system design.

Figure 2 is a typical all waveguide diplexing, switching and filtering system.

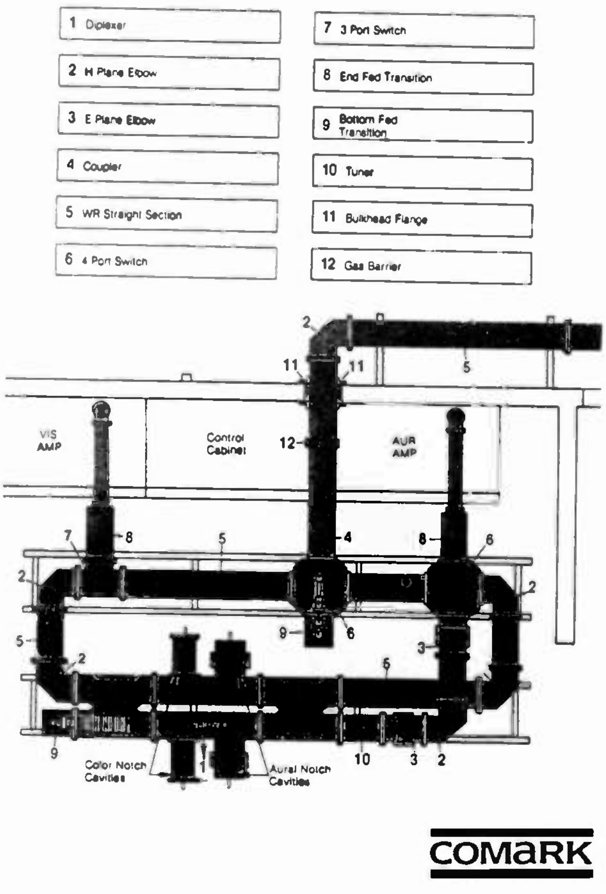

Figure 2 - Typical Internal Waveguide System Installation

(*) Visual system power rating includes a factor of 10% Aural.

Figure 3 is a typical waveguide RF power transmission system to the antenna.

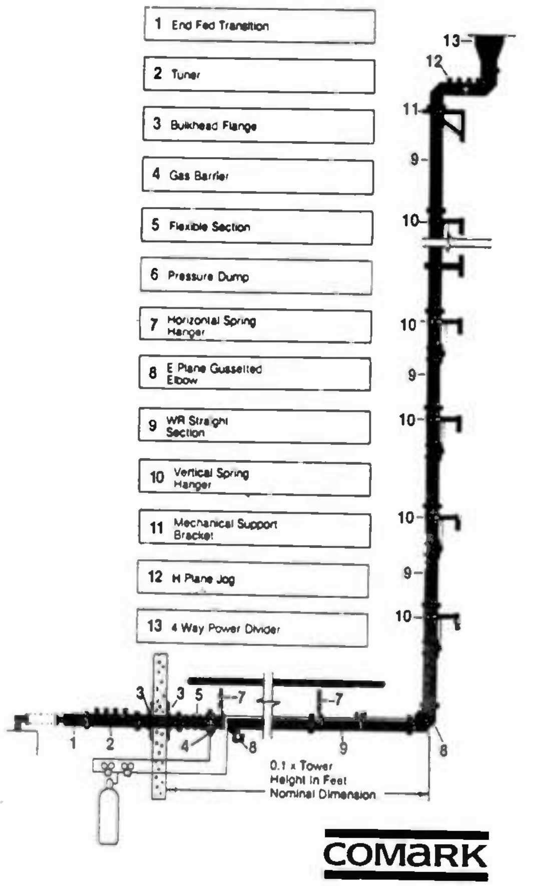

Figure 3 - Typical External Waveguide System Installation

C. The Transmitter

The advantages of super power have not gone totally unnoticed by the broadcast industry. However, the economics of operating a 300kW to 400kW transmitter have prohibited widespread implementation. There are several 220kW to 240kW stations operating today taking some advantage of the super power concept. Comark has built two of these stations rated for at least 220kW transmitter output power.

Significant advances have been made in recent years to improve klystron efficiency[1] . New tubes and new pulsing techniques have made the 200kW power level economically feasible. However there seems to be no immediate and simple solution that will permit klystrons to become efficient enough to operate at 300kW to 400kW economically.

The possibility of achieving super power then must rely on new technology. This new technology is now available.

Several papers have described the efficiency of the klystrode device[2,3,4]. A production transmitter has now been manufactured by Comark which uses a 60kW tube and a 600W solid state driver. Figure 4 is a photograph of the klystrode final amplifier under test in Comark's factory.

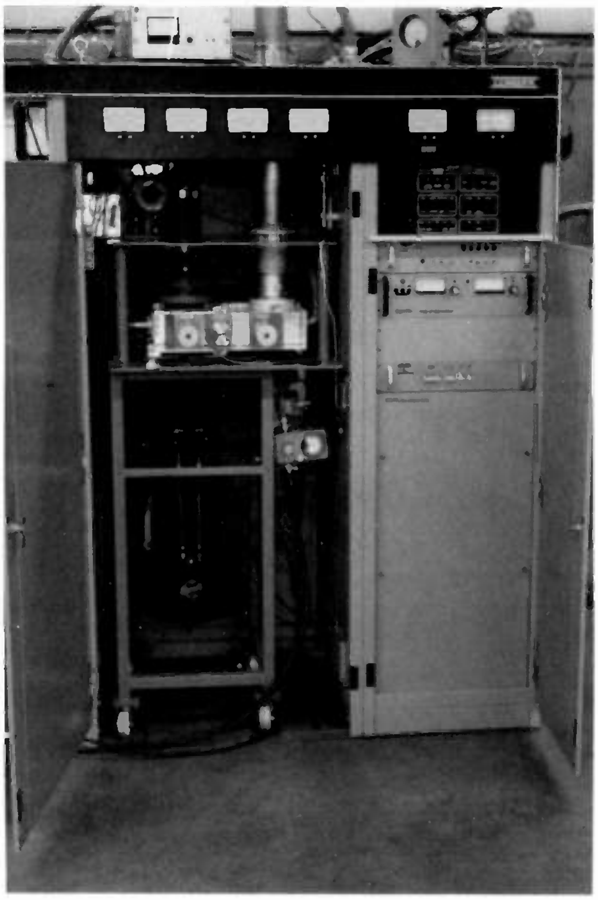

Figure 4 - The Klystrode final amplifier under test in Comark factory

Detailed factory tests indicate that the transmitter is not only desireable and efficient at 60kW, but is a likely candidate for a building block to super power levels in excess of 350kW.

Table I is an example of power consumption comparisons.

	Peak Output Power	Avg. Beam Power 50% APL	Figure of Merit*	Advantage Ratio
Klystrode	60kW	46.5kW	1.29	1.87
Pulsed Klystron	60kW	86 kW	0.69	

* Peak RF Output divided by Average Beam Power

The Class B Klystrode relies on the physics of its internal structure to continuously vary beam current directly with RF drive. The pulsed klystron can only reduce beam current during the period between sync pulses and then only to black level power. A measured 1.87 advantage of the klystrode over the klystron exists in power consumption during average picture conditions. Measurements on the Comark 60kW klystrode transmitter indicate that even at full black and sync drive levels the klystrode's beam efficiency is no worse than the best pulsed klystrons (i.e. 70%).

With this enormous energy consumption advantage it appears that a 350kW super power klystrode equipped station would have a lower electric bill than a 240kW klystron powered station. A 120kW klystrode powered station would approach the power cost of a 60kW pulsed klystron installation.

Figures A through E show the measured linearity performance of Comark's 60kW klystrode powered UHF amplifier, driven by the Comark system exciter and operating at full power. At 50% APL the beam voltage was 31.5kV and beam current was 1.45 Amps for 61kW peak RF output power.

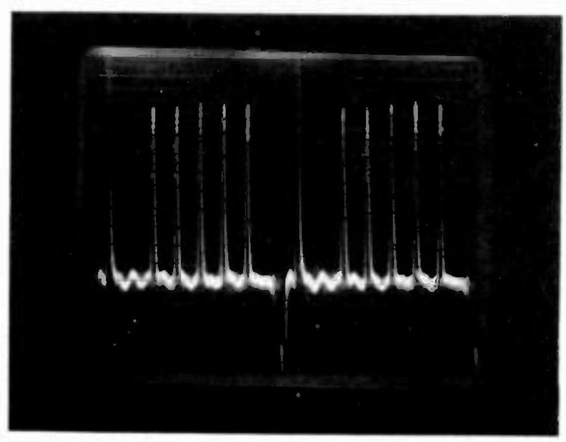

Figure A - Klystrode Transmitter Output - Low Frequency Linearity at 61kW Peak Sync

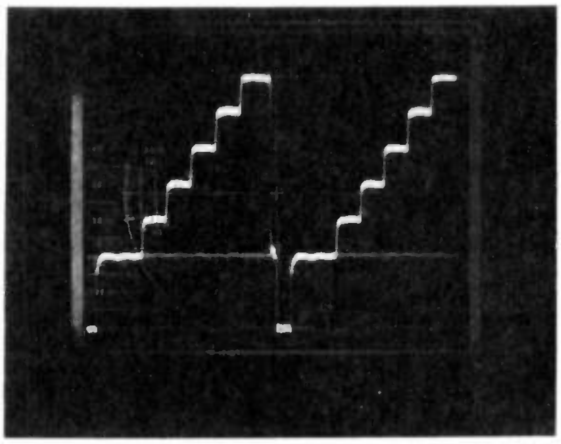

Figure B - Klystrode Transmitter Output - Stairstep at 61kW Peak Sync

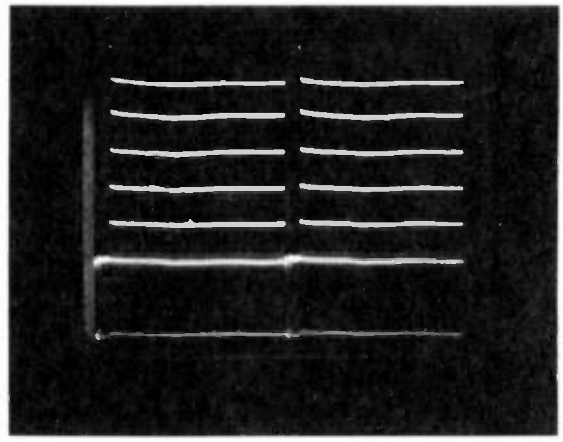

Figure D - Klystrode Transmitter Output - Field Rate Stair Step

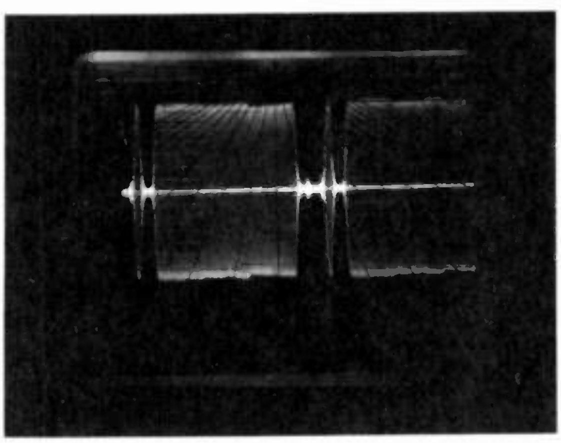

Figure C - Klystrode Transmitter Output - Differential Gain - 61kW Peak Sync

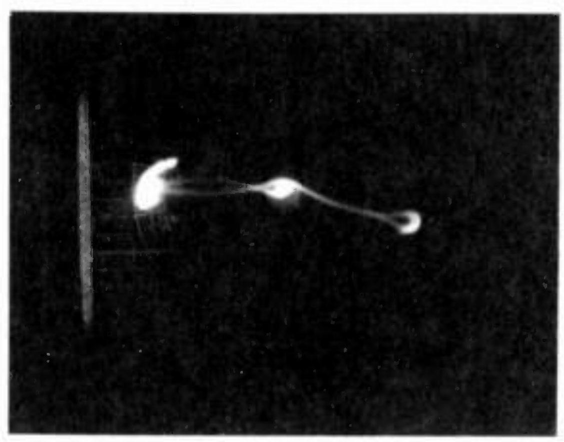

Figure E - Klystrode Transmitter Output - Differential Phase - 61kW Peak

The construction of a super power transmitter requires the combining of multiple output amplifiers. This multiplicity of parallel stages not only provides the required RF energy but also improves overall station reliability.

Proper distribution of D.C. power supplies, cooling systems and RF drive sources can virtually guarantee continuous on-air service.

Comark has configured four 60kW klystron finals using multiple cooling and power supply units in combination with RF switching to produce 240kW output powers with high on-air reliability. Phasing and balance problems in the four-way combination were not difficult to control, even under deeply pulsed klystron conditions. The use of waveguide switches and automatic control logic permit the selection of several combinations of RF drive and output amplifier paralleling geometry.

The experiences gained from four klystron, 240kW transmitter designs and their operation can be quickly applied to the higher levels of amplifier combination and output power required for super power operation.

IV. CONCLUSIONS

This paper has presented a concept which could be called "UHF Super Power." The concept of super power operation is proposed as a result of the significant advantage gained in the city grade coverage area of a station by the use of low horizontal gain, wide vertical beam, transmitting antennas.

Field strengths in excess of 10db to 20db over today's narrow vertical beam, high gain, antennas can be realized with this technique. The need for this higher field intensity has been created by the competitive effects of alternative video program delivery systems that do not rely on over the air technology.

The low gain antenna requires much higher transmitter output power. Until the introduction of the Klystrode, the economics of power consumption limited transmitter outputs to 240kW. The klystrode, however, has been demonstrated to have significantly higher energy efficiency. In fact, a 350kW klystrode transmitter could cost no more to operate than a 240kW klystron equipped transmitter.

The high power required to benefit from Super Power operation also defines the antenna and transmission line configurations.

Fortunately, waveguide transmission system technology is well developed and quite capable of handling the RF power levels associated with Super Power.

The final technological element necessary to make Super Power possible was the development of the waveguide antenna. This device does not rely on an individual power splitter or phasing harnesses to feed the antenna's radiating elements. It relies on the waveguide structure itself. Thus, the reliability of the antenna approaches the reliability and power handling capacity of the waveguide transmission line feeding the antenna.

Two new developments, the klystrode transmitter and the waveguide antenna, combined with an existing technology, waveguide transmission systems, are the three elements which come together to permit the UHF broadcaster to respond to the challenge presented by non over-the-air program sources.

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ADAPTING WIDEBAND EXTERNAL CAVITY KLYSTRON TECHNOLOGY TO INTEGRAL CAVITY TRANSMITTERS

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ABSTRACT

This paper details the research, rationale, and process of modification for replacement of integral cavity klystrons with external cavity klystrons in existing transmitter systems. Criteria for successful replacements are set and reached. The physical modification process is detailed and finally, examples of other replacements are given. Two actual conversions that have been performed and that are presently operating are described.

INTRODUCTION

Media Central Inc. is a group owner and management company of independent UHF-TV stations. The company has built and managed 10 stations within the past 5 years, involving 21 klystron sockets (19 integral versions) with a total of 23 failures during this same period. Only two of these failures could be directly attributed to either an operator error or transmitter malfunction. The balance of these failures included out-of-the-box initial failure, loss of emission, loss of current, loss of vacuum, mod anode short, cavity tuner leaks, etc. Media Central, as a management entity, has become painfully aware of the cost of downtime created by these failures and the replacement cost of each klystron, even while under the pro-rata warranty. In contacting a number of stations in the industry, it became apparent that our situation is not unique.[2]

Due to the high incidence of failure with the integral cavity klystron during the past four years, and assessing the cost of replacement coupled with the other attendant advantages of the external cavity klystron, it was determined that one solution would be replacement of the internal cavity klystron with the recently developed "wideband" or "all channel" external cavity klystron. The first such exchange was accomplished successfully in January of 1986, and a second unit was converted in April of the same year. Both systems have been operating reliably with no downtime. The process has thus been proven successful and is an alternative that should be considered by every UHF TV engineer who must make a decision on the purchase of their next klystron.

Research on new technologies, such as the Multistage Depressed Collector klystron or the klystrode may present even more attractive solutions in the future, but are currently in the laboratory development stage. The method of using the external cavity klystron is available for immediate implementation and the transmitter retains FCC type acceptance. Figure 1 is the front view of a completed installation.

BASIS AND RATIONALE FOR REPLACEMENT

The predicted normal useful life of a klystron is in the neighborhood of 30,000 to 35,000 hours of beam operation. The aggregate time between failures is defined by the following formula provided by the manufacturer and is known as the Mean Time Between Failures (MTBF):

$$\text{MTBF} = \frac{\text{Failed Hours} + \text{Working Hours}}{\text{Number of Failures}}$$

where: Failed hours = Cumulative hours of all failed prior to this failure

Working hours = Cumulative hours of all presently operating units

There are many klystrons that have met this life expectancy and have indeed exceeded it by large margins. Media Central's experiences have not been so fortunate, yielding a much lower figure of 12,600 hours.[3] Recently, other stations have only been experiencing 18-23,000 hours.[4]

As can be seen, the Media Central track record was far below the predicted level and considerably below the sampled averages mentioned above. Frequent failure of klystrons not only affected the amount of downtime generated during the failure, but also the streamlined budgets that our stations are designed to operate within. A great deal of time and effort was spent troubleshooting the transmitters in hopes of finding and external cause for the

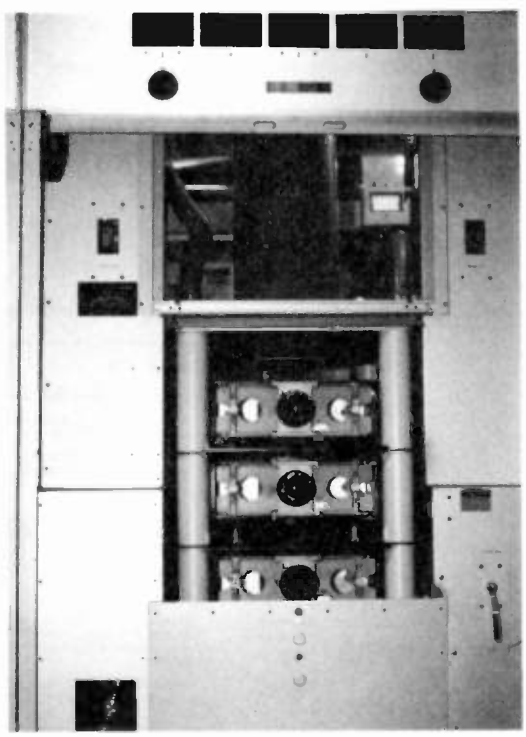


FIGURE 1

failures. Although improvements were made on certain design areas, nothing was found that would directly cause the type of failures that we were experiencing. The klystron manufacturer was the most cooperative in providing assistance during this time, as was the transmitter manufacturer, RCA. Still, with all the gathered data and improved modifications, Media Central stations experienced continued failures.

CRITERIA FOR REPLACEMENT

In light of this experience with Integral cavity klystrons, an alternative power amplifier was sought. Before any replacement could be implemented, the following conditions had to be met:

1) **RELIABILITY:** A proven track record was necessary to insure that the only variable to contend with was the installation process, not the power amplifier performance.

2) **FUTURE AVAILABILITY:** A design that will be available for the foreseeable future.

3) **MULTIPLE SOURCES:** Not tied to a single supplier.

4) **WORKING ASSEMBLIES:** The alternative had to be a fully developed technology. Prototyping was unacceptable.

5) **COST EFFECTIVE:** The alternative had to be economically feasible.

6) **READY STOCK:** Replacement units must be immediately air shippable.

7) **WARRANTY:** Alternative must supply equal or better warranty time.

8) **NO DOWNTIME DURING CONVERSION:** It was highly desirable to be able to modify the transmitter without losing airtime.

9) **TRANSPARENCY:** Installation must be compatible with the present system structure, maintain the same control and safety functions and meet or exceed the present proof of performance data.

10) **FCC COMPLIANCE:** The system must comply with all aspects of the FCC rules.

Additional advantages identified that were relevant to the external cavity klystron were:

11) **ABILITY TO PULSE:** Conversion to pulsing techniques at moderate to low cost.

12) **CHANNEL INTERCHANGEABILITY:** Ability to stock one klystron and one magnet assembly common to all stations.

A series of telephone calls were made to klystron manufacturers and UHF TV stations in order to evaluate presently existing systems. It developed, in the opinion of Media Central, that the higher marks were given to external cavity klystrons as the best alternative power amplifier for the transmitters involved. Although it meant sacrificing certain features of the integral cavity klystron, such as rapid replacement, Media Central made the decision to replace the integral cavity klystron with an external cavity klystron. Much attention was given to the present state of research in klystron technology including the Multistage Depressed Collector klystron and the re-emerging efforts with the klystrode, but these failed our criteria on the grounds of lack of a track record and availability of commercially working units.

In short, by the process of elimination, the "wideband" external cavity klystron became the only real alternative to Media Central's problem.

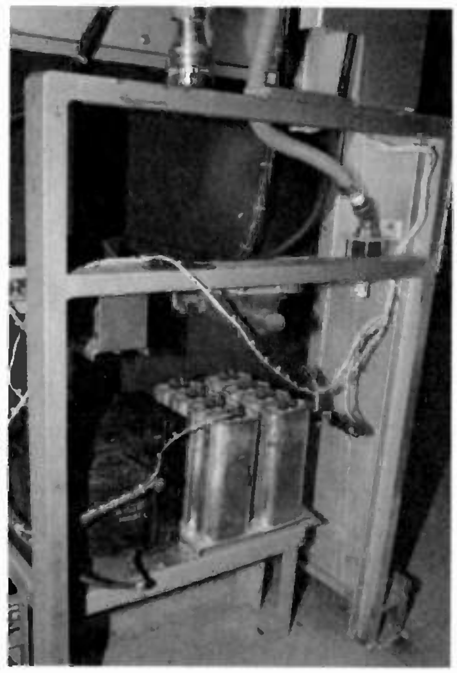


FIGURE 2

	EXTERNAL	INTEGRAL
Initial Cost	Similar	Similar
Spares Cost	Cheaper. Only beam tube wears out.	Dearer. Beam tube and therefor cavities wear out.
Ease of Replacement	Straightforward. No distrubance of state of tune of cavities required. Quite rapid.	Straightforward. Very fast provided no retuning required.
Tuning	Easy and versatile. Optimum efficiency available because loading on all cavities can be varied.	Usually supplied factory pre-tuned therefore easy. Otherwise not easy and optimum efficiency not achievable because loading cannot be varied on all cavities.
Tuning Range and Effect on Spare Stock	One tube and magnetic circuit assembly to cover 470-860 MHz offers networks possibility of reducing spares holdings.	Three tubes and magnetic circuit assemblies to cover 470-860 MHz. Larger spares stocks required to maintain adequate back-up.
Efficiency	Good over ALL channels.	Good at some channels, poor at others. With newly developed variable output couple good at all channels but at additional initial expense.
Operating Environment	Known to operate satisfactorily in all climates. No special air filtering necessary beyond that normally required for ventilating the rest of the transmitter cabinet.	Small bore pipes used for cavity and body cooling mean that water quality has to be carefully controlled and safeguards taken against freezing of coolant left in the pipes when not in use.
Size	Latest designs require 510mm X 510mm floor space up to 30 KW and 546mm by 546mm 40KW to 64KW.	Cabinet volume required similar to new external systems since the latter were designed to fit the same transmitter housings.

The above chart is taken from the EEV pamphlet on UHF Television entitled "External or Integral Cavity System". Used with permission.

DETERMINING THE SUSCEPTIBILITY OF THE TRANSMITTER TO MODIFICATION.

The transmitters subject to modification were Media Central's eight RCA TTU 55's and one RCA TTU 110C. A study was conducted to assure that the transmitters could be properly modified. The following items were examined:

1) **POWER REQUIREMENTS:** Various power supplies would have to be modified to accept the differing voltages of the external cavity klystron.

   A) **Beam Supply:** Electro Engineering Works unitized beam supply as furnished with the transmitter is capable of 23.5 to 26.5 KVDC if needed. No change was necessary.

   B) **Magnet Supply:** Lower current demanded by the new klystron, reused existing supply. The choke and filter capacitors were moved. See Figure 2.

   C) **Filament Supply:** Integral klystron requires 7 VAC at 16.5 to 17.5A. External klystron requires 8.5 VDC, regulated, at 28.5A. Filament supply was replaced.

   D) **Mod Anode:** No component changes, existing resistor ladder was used for the necessary taps.

   E) **Ion Pump:** Not required in the existing circuit, but highly recommended and implemented in each conversion.

2) **CONTROL CIRCUITS**

   A) **Interlocks:** Magnet and body water flow interlocks are no longer required. Airflow interlock added to facilitate new magnet assembly. Physical interlocks for old magnet assembly removed.

   B) **Filament:** New two step time delay for filament protection, connected to interlock ladder and TD/Bypass switch interfaced to insure klystron protection. Figure 3 shows the new assembly.

3) **RF**

   A) **Increased input drive:** Provided by the addition of an intermediate power amplifier capable of delivering 30 watts of RF.

   B) **Interface:** A run of 3⅛" unflanged transmission line was used to link the klystron output to the existing harmonic filter. A special adapter was constructed to match the 3⅛" line to the custom inner conductor bolt of the dual harmonic filter assembly. The new configuration provides the ability to multiplex from either klystron assembly with ease.

4) **WATER SYSTEM**

   A) **New top hat boiler interface:** Constructed to maintain the integrity of the vapor phase cooling system. See Figure 4.

   B) **Water drain:** Relocated to provide access for front entry of the klystron dolly.

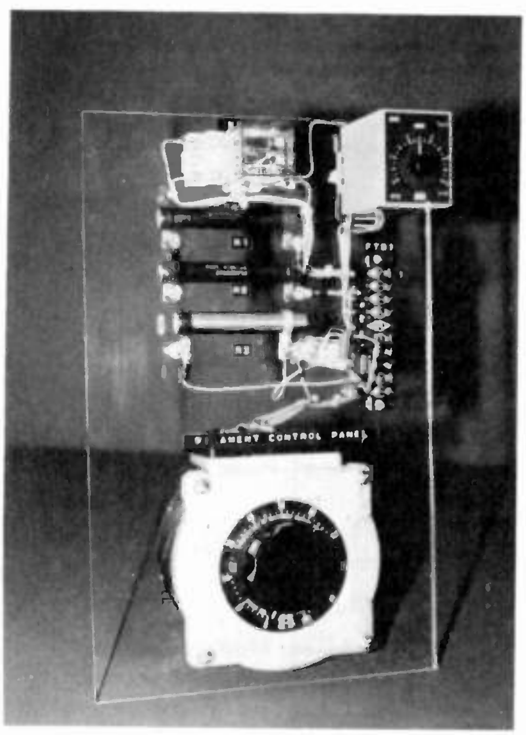
   C) **Magnet and klystron water supply:** Plugged and bypassed.

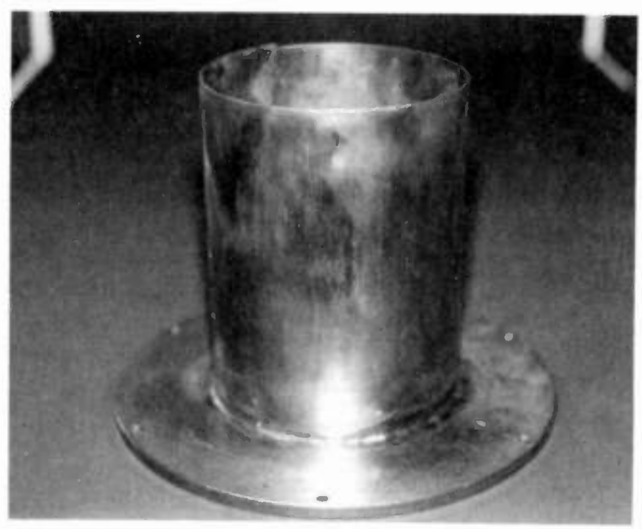
IMPLEMENTATION

A key to the success of the conversion was planning and timing. The added experience of having constructed and closely supervised various transmitter installations helped a great deal in the prefabrication of the required components for the conversion. During the process, the transmitter is not disabled, it can continue normal operation in the multiplex mode. Significant structural and electrical changes occur during scheduled late night off air hours. The time to completion, upon arrival on site (excluding Proof of Performance) is three days.

Prior to conversion, Willard Shears, Director of Engineering, WNVC-TV was contacted. He had converted an older style external cavity circuit to the new "wideband" style with the addition of pulsing. His conversion was successful and provided useful background.

The first conversion took place at WOAC-TV67, Canton, Ohio, with the assistance of Lee Carpenter, Chief Engineer.





The aural klystron was failing rapidly and the system was already in multiplex pending its repair. The filament control circuit and supply were pre-built and all items necessary for the conversion were acquired prior to arriving on site. The actual conversion took place over three nights. A delay was incurred subsequent to the installation due to faulty welding in the construction of the boiler top hat. An ITS 15 watt IPA was used due to the lower drive level required for aural operation. The circuit was outfitted for operation as either aural or visual and, even with the lower drive, the circuit was tested in visual service at 80%. The system, once completed, performed from the moment it was turned on. The Proof of Performance was made of the system and the only appreciable change was an unexplained drop in the VSWR from a previous 1.12 to a present 1.06. The total time spent on site in January 1986 was 8 days.

The second conversion took place at WKCH-TV 43, Knoxville, Tennessee, with the assistance of Lamar Gilbert, Chief Engineer. The visual integral cavity klystron had suffered a cracked output window. Again, all components were acquired and prefabricated prior to arriving on site. This time, however, building on prior experience, the total installation and Proof of Performance involved only four days on site. The visual external cavity circuit was installed in the aural circuit, while the remaining aural klystron was placed into the visual socket and used for multiplexing. Upon completion of the installation, the new external cavity klystron was placed into visual service while the integral cavity klystron was re-routed into the aural output line. All of the meter functions remained the same, although the aural and visual sides were swapped. Figure 5 shows the front view of the WKCH-TV installation with Integral and External klystrons side by side for comparison. A block diagram is at Figure 6.

RESULTS

A comparison with the criteria previously set forth shows these results:

1) **RELIABILITY:** As of this writing, no failure or downtime has occurred in either installation due to the conversion or associated circuitry. WOAC-TV is approaching 8500 hours in aural and WKCH-TV is approaching 6700 hours in visual.[5] both stations operating a minimum of 18 hours a day.

2) **FUTURE AVAILABILITY:** The new series of "wideband" external cavity klystrons represent the latest technology in design improvements over the older band selective series.

3) **MULTIPLE SOURCES:** EEV and Amperex provide totally interchangeable klystrons and magnet assemblies, with complete compatability in performance and price. Thorn EMI/Varian is introducing its version as a direct substitute for the EEV and Amperex. This provides at least three sources, all with promised stock in the U.S.

4) **WORKING ASSEMBLIES:** Systems have been performing successfully in the marketplace for over four years.

5) **COST EFFECTIVE:** Even though initial conversion costs are slightly higher than direct replacement of

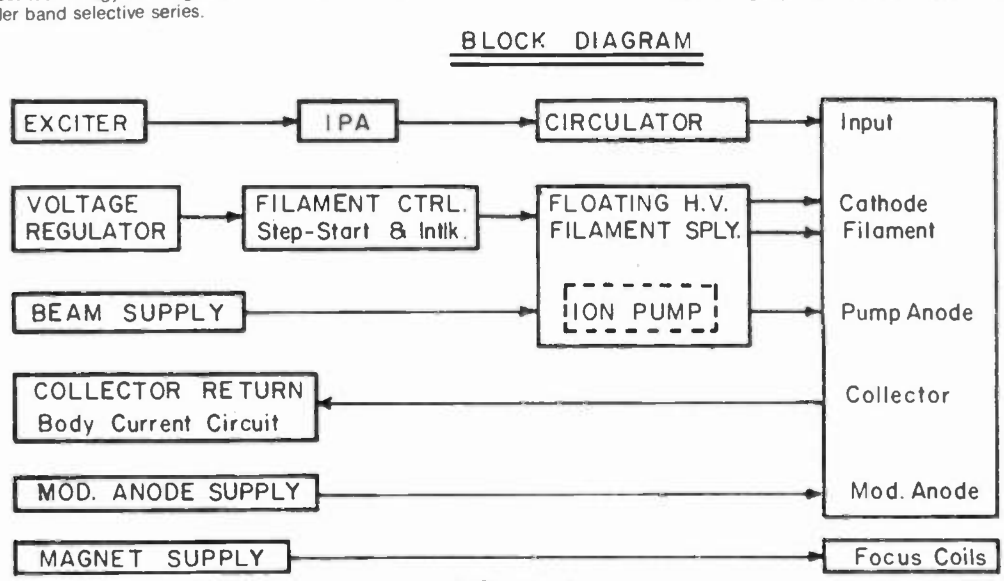


FIGURE 6

an integral cavity klystron, the costs are quickly repaid in operating expense savings, possible longer life, and a replacement cost of half of the present integral version.

6) **READY STOCK:** Both EEV and Amperex have a ready stock of klystrons and magnet assemblies ready for twenty-four hour shipment. Under normal circumstances, no prior tuning is necessary as only the beam tube is replaced.

7) **WARRANTY:** Warranties are available for 10,000 hours, on a pro-rata basis.

8) **NO DOWNTIME DURING CONVERSION:** As previously mentioned, normal operation is continued in multiplex operation.

9) **TRANSPARENCY IN INSTALLATION:** The real advantage for the transmitter engineer is the transparency or equivalence of operation of the modification. Aesthetically, the change of the outside appearance is minimal, still remaining clean and neat. The sequence of controls for remote operation is not affected. Meter readings and maintenance procedures are routine, with fewer water flow meters. A plus to the tuning procedure is the ability to more closely tune and match the klystron to the RF chain. There is no need for external high efficiency or variable visual couplers.

10) **FCC COMPLIANCE:** None of the critical operating parameters of the transmitter is changed from that authorized in the type acceptance by the FCC. Mass Media Docket Number 86-264 permits modification of a transmitter which is on the FCC's "Radio Equipment List" without FCC notification or authorization. The FCC now only requires that equipment performance measurements be made within ten days after completing the modifications and that an informal statement or diagram describing the modification be retained at the transmitter site for as long as the equipment is in use.

11) **ABILITY TO PULSE:** Amperex's Annular Beam Control, EEV's Beam Control Device and Varian's Annular Control Electrode provide a common pulsing technique that is retrofittable at approximately half the cost of the circuit required for the integral version.

12) **CHANNEL INTERCHANGEABILITY:** For the two previous installations and any future installations, Media Central may now stock a single model klystron capable of performing on all UHF channels. This offers a unique advantage to multiple station owners not economically feasible in the past. The integral and older external cavity klystrons come in three distinct channel bands and sizes, and the proposed klystrode has cut the number to only two models to cover the band.

The conversions in the Media Central Inc. stations incorporated the Amperex YK1265, primarily due to the successful experience with our PYE-TVT 30KW klystrons. Willard Shears, with WNVC-TV, also used the Amperex YK1265 in his conversion from the older and larger external cavity assembly. Leroy Wallace of SMD Incorporated upgraded an RCA 30KW UHF to a 60KW using the EEV K3672 BCD, a compatible assembly to the Amperex YK1265. SMD is also in the process of completing a 110KW in an RCA transmitter using the same EEV klystron.

SUMMARY

The undertaking of this conversion process is not difficult in itself. The details, however, are many and require thorough attention. The conversion, along with a new solid state exciter, can upgrade the performance of an older transmitter presently being considered for retirement, and turn it into a reliable, state of the art transmission system at a fraction of the cost of a new transmitter.

Replacing integral cavity klystrons with external cavity klystrons is not new. In 1980, EEV began replacing integral cavities with externals in Finland and modified 12 sockets over a period of two years. Subsequently, more modifications were made in Yugoslavia. These retrofits were made using the older "three-across-the-band" external cavity klystrons in both RCA and Gates transmitters. EEV has also converted two NEC visual sockets in Cairn Hill, Erie, Ireland using the "wideband" klystrons discussed in this paper.

The time has come for engineers to exercise their abilities and creative expertise to do their part to keep costs down and reliability up in their stations. As we all become keenly aware of the importance of "the bottom line", the effort to cut down on tube expenses and operating costs becomes welcomed by management, who seldom give such credit to engineering.

FOOTNOTES AND CREDITS

1) ***Matthew A. Sanderford Jr.,** B.S.E.E., University of Texas at El Paso, Director of Engineering, Media Central, Inc. Direct questions to Mr. Sanderford at 620 Osborne Office Center, Chattanooga, Tenn., 37411. 615-899-7920.*

   *Assistance in the preparation of this article is acknowledged of **Rick G. Morris,** B.A., University of Illinois, J.D., University of Kansas, Chief Engineer, KZKC 62, a Media Central station.*

2) *It is not the intention of this article to join in the debate on industry-wide averages in reliability and power consumption, that is left to the independent testing agencies and the manufacturers who have been debating relative reliabilities for years. Rather, this article is meant to document one group owner's experiences and provide detailed information on one solution for evaluation by peer engineers for possible application in their stations. Reliability was of importance in assessing the desirability of the conversion process, but the only claims of reliability made herein are those as actually documented at Media Central stations. Other considerations involved in deciding to convert were the extremely low cost of replacement and the possibility of only stocking one spare for the group. Each such factor will have to be weighed by the engineering talent on site when making the decision on whether to convert.*

3) *As of February 1, 1987*

4) *Gathered from informal conversations with field service engineers.*

5) *As of April 1, 1987.*

KLYSTRON MANUFACTURERS:

1) Amperex, 230 Duffy Ave., Hixville, N.Y. 11802
   Pete Fochl 516-931-6208

2) EEV Inc., 4 Westchester Plaza, Elmsford, N.Y. 10523
   Mike Kirk 1-800-431-1230

3)) Thorn EMI/Varian, 611 Hansen Way, Palo Alto, Calif. 94303
   John Ahorn 415-493-4000

The author gratefully acknowledges the assistance of the above companies who have provided information, klystron data and other assistance for the conversions and this paper.

PHOTO CREDITS

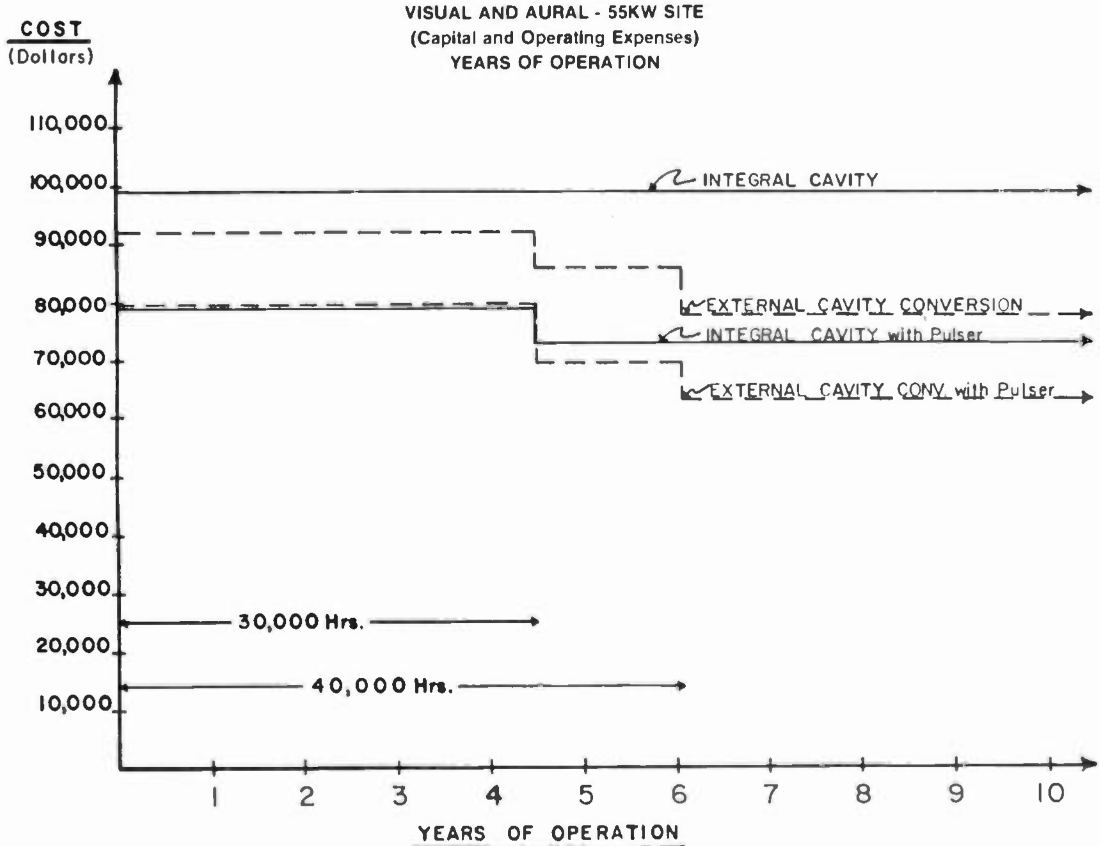
Photographs were furnished by Lamar Gilbert and Matthew Sanderford, Jr.

MEDIA CENTRAL INC.
External Cavity Klystron Conversion

COST ANALYSIS

.. were used from manufacturer predicted performance and industry averages. They were applied uniformly in ..rough 4. Item 5 reflects performance actually experienced during the past 5 years working with integral cavities in 7 ..nitter installations, and 1.3 years with external cavity conversions. Average daily meter readings were used to compute the ..opriate power requirements for each circuit.

TYPICAL COST OF OWNERSHIP
VISUAL AND AURAL - 55KW SITE
(Capital and Operating Expenses)
YEARS OF OPERATION



1. **INTEGRAL CAVITY (no modifications)**
   CAPITAL: Visual $12,035./yr.
   Aural $ 9,016./yr.
   OPERATING: Visual + Aural = $78,780./yr.
   0 to 10 yrs. — $99,831./yr.
   Total cost of ownership for 10 years .... $998,310.

2. **INTEGRAL CAVITY (with Pulser and Visual Variable Coupler)**

CAPITAL:	Initial	Replacement		
Visual	$18,052./yr.	$12,035./yr.		
Aural	$ 9,016./yr.	$ 9,016./yr.		
OPERATING: Visual + Aural = $51,647./yr.				
0 to 4.57 yrs.	— $78,715./yr.		for 4.57 yrs.	$359,728.
4.57 to 6.1 yrs.	— $72,698./yr.		for 1.53 yrs.	$394,750.
Total cost of ownership for 10 years				$754,478.

3. **EXTERNAL CAVITY CONVERSION**

CAPITAL:	Initial	Replacement		
Visual	$14,223./yr.	$ 6,127./yr.		
Aural	$10,655./yr.	$ 4,590./yr.		
OPERATING: Visual + Aural = $67,145./yr.				
0 to 4.57 yrs.	— $24,878./yr.		for 4.57 yrs.	$420,545.
4.57 to 6.1 yrs.	— $83,927./yr.		for 1.53 yrs.	$128,408.
6.1 to 10 yrs.	— $77,862./yr.		for 3.9 yrs.	$303,662.
Total cost of ownership for 10 years				$852.615

4. **EXTERNAL CAVITY CONVERSION (with Pulser)**

		Initial	Replacement
CAPITAL:	Visual	$16,411./yr.	$ 6,127./yr.
	Aural	$10,655./yr.	$ 4,590./yr.

OPERATING: Visual + Aural = $52,750./yr.

0 to 4.57 yrs.	— $79,816./yr.	for 4.57 yrs.	$364,759.
4.57 to 6.1 yrs.	— $69,532./yr.	for 1.53 yrs.	$106,384.
6.1 to 10 yrs.	— $63,467./yr.	for 3.9 yrs.	$247.521.

Total cost of ownership for 10 years $718,664

5. **MEDIA CENTRAL INC. rationale for conversion.**
As of 2-1-87, Mean Time Between Failures ,(MTBF), was approx. 12,600 hrs. This is comprised of 15 integral circuits with a total of 23 failures. 12,600 hrs. = 1.92 yrs. actual life. Equal failure rate occurred for both Visual and Aural units. Comparing present rate to expected conversion life:

Actual Cost to Media Central to Operate Integral Cavity Klystron

CAPITAL: Visual $28,646./yr. (for 1.92 yrs.)
Aural $28,646./yr. (for 1.92 yrs.)
OPERATING: (from item #1) = $78,780./yr.
Cost per year — $136,072./yr.
Total cost of ownership for 10 years .... $1,360.720.

Projected Savings using External Conversion

0 to 4.57 yrs.	— $44,049./yr.	for 4.57 yrs.	$201,304.
4.57 to 6.1 yrs.	— $50,145./yr.	for 1.53 yrs.	$ 76,722.
6.1 to 10 yrs.	— $58,210./yr.	for 3.9 yrs.	$227,019.

Total owner savings over 10 years .... $505,045.

Projected Savings using External Conversion with Pulser

0 to 4.57 yrs.	— $56,256./yr.	for 4.57 yrs.	$257,089.
4.57 to 6.1 yrs.	— $66,540./yr.	for 1.53 yrs.	$101,806.
6.1 to 10 yrs.	— $72,605./yr.	for 3.9 yrs.	$283,160

Total owner savings over 10 years .... $642,055.

Notes:

1. Predicted Visual life - 30,000 hrs. (4.57 yrs.)
2. Predicted Aural life - 40,000 hrs. (6.1 yrs.)
3. Typical broadcast day - 18 hrs. (Beam power usage)
4. Electrical Power cost - $0.07/KW/hr.
5. Operating Cost Factor - .07 x 18 x 365 = $459.9/KW/Yr.
6. CAPITAL costs include the cost of purchase for the klystron and all expenses incurred to modify the circuit klystron and all expenses incurred to modify the circuit from the basic supplied integral configuration. Included also are the shipping and additional charges required in some instances to insure delivery within a reasonable 72 hr. period from failure to recovery. All related costs are amortized at the same rate as the life of each klystron.
7. OPERATING costs were based on actual power consumption of one Visual and one Aural, 55KW configuration, plus the additional power from directly associated circuits. These costs remain constant for each configuration as long as that type circuit remains in use.

RECENT DEVELOPMENTS IN FIVE-CAVITY UHF-TV KLYSTRONS

Earl W. McCune
Varian Associates, Inc.
Palo Alto, California

INTRODUCTION

UHF-TV stations are particularly concerned about power consumption of the power amplifier. The Varian VKP-7553,4,5 series of klystrons (called the S-Models) are designed to optimize efficiency performance. The design considerations were described previously (1) but can be summarized as arranging the rf circuit and cavity tuning in a manner to enhance the rf current on the electron beam. A five cavity klystron is required to allow the efficiency enhancement while maintaining the gain and bandwidth performance necessary for TV service. Tubes of this design typically provide 52% saturation efficiency. The VKP-7553S klystron is shown in Figure 1.

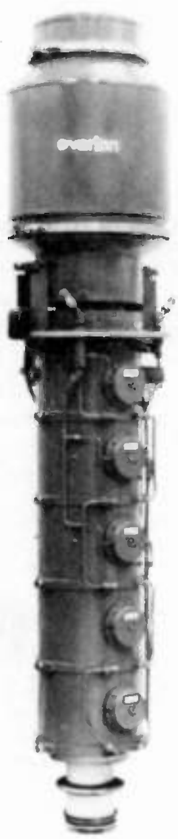


FIGURE 1. VKP-7553S KLYSTRON

Varian has undertaken a development program to refine the "S-Tube" design by improving the electron gun performance. An upgraded gun design has been achieved which provides an improved beam allowing increased beam-to-cavity-coupling resulting in improved rf performance. The new gun design incorporates an insulated focus electrode to allow beam control with lower voltage then required by the modulating anode. The beam control electrode (BCE) allows use of advanced beam pulsing techniques to optimize overall transmitter efficiency. The new klystron is designated the "SE Klystron".

Also described is another program in progress to develop a control grid design with high amplification factor to allow maximum efficiency enhancement using low voltage beam modulation techniques.

ELECTRON GUN DESIGN

The electron gun refinement program has two objectives. The first is to reevaluate the existing design using modern gun design technology and incorporate modifications as necessary to achieve an optimum design. The second objective is to incorporate electrical insulation of the beam focusing electrode to allow beam current modulation by this electrode. This program has been successfully completed. Tubes incorporating the new electron gun design demonstrate improved rf performance.

This program used the Varian beam analysis facility which can provide a detailed determination of electron beam performance. The facility is used for final refinement of electron gun designs, generally after preliminary design is accomplished by computer simulation techniques. This two-step design process results in electron gun designs that provide beams of excellent laminarity. Beam analyzer results for the new design show the current density across the beam sampled along the axis in Figure 2. We can see that the beam edge is sharply defined and the current density is enhanced slightly at the beam edge. This is typical for this type of gun design and is useful to provide better beam to cavity coupling. Computer processing of the analyzer data provides the presentation of Figure 3 giving a perspective view of the beam cross section.

The beam control characteristics of the beam control electrode have been determined as shown in

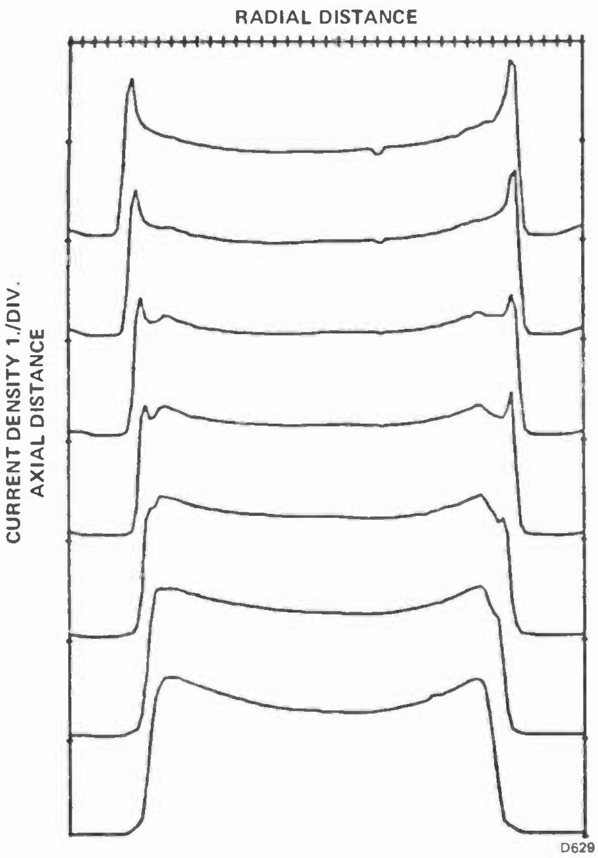


FIGURE 2. BEAM PROFILES

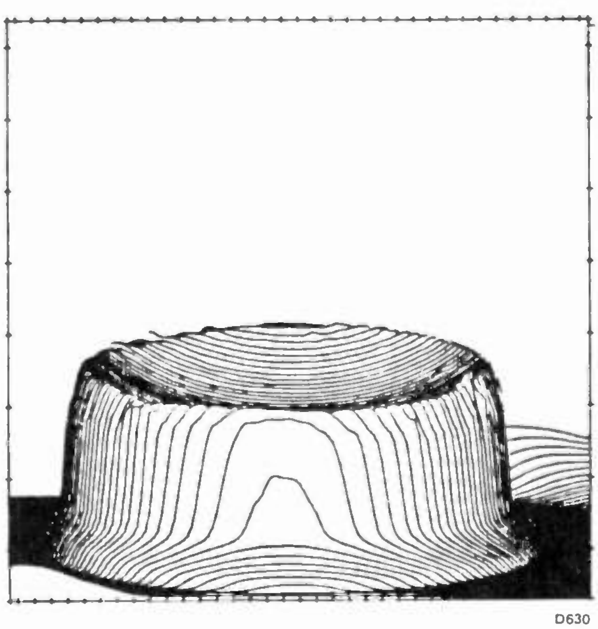


FIGURE 3. PERSPECTIVE VIEW OF BEAM PROFILE

Figure 4. Typical operation with a 60 kW klystron would be with the cathode at -24.5 kV and the modulating anode at 18 kV with respect to the cathode (-6.5 kV with respect to ground). The BCE would be at cathode potential for full sync pulse

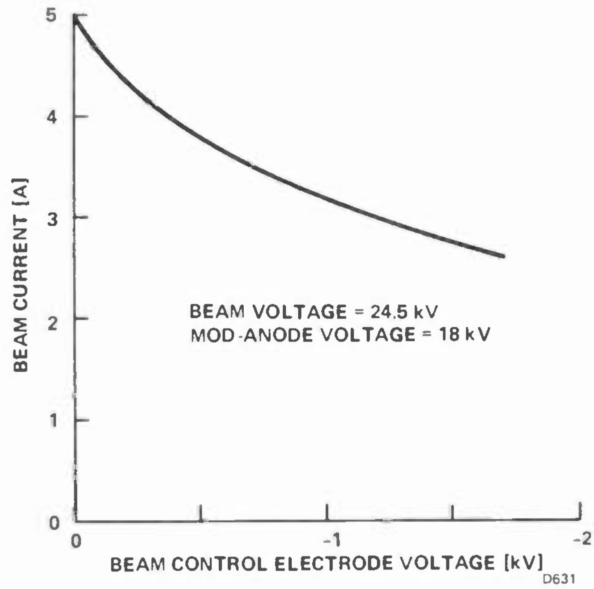


FIGURE 4. BEAM CONTROL CHARACTERISTICS

output and at -1000 V for blanking level output. The beam current is reduced from 4.85 A to 3.0 A for these conditions.

KLYSTRON PERFORMANCE

Klystron performance using the new design is shown in Figure 5. The gain characteristics (output power versus rf drive power) are shown for beam current levels appropriate for sync pulse level and blanking/picture levels. The inherent efficiency capability is demonstrated by the maximum value of 56% at rf saturation at the sync power level. Efficiency remains high, however, even at the reduced power level. With 3.15 A beam current, the efficiency is still over 50%. For these data, the tube was adjusted for optimum loading by using the variable load coupler at the sync power level. For comparison, Figure 6 shows similar data for a typical S-Klystron. The new design has enhanced efficiency from 53 to 56%, representing a 6% improvement. The BCE has reduced beam modulator voltage requirements from 5000 to 700 volts. The reduced pulsing voltage will allow use of solid state modulators with improved capabilities.

ADDITIONAL DEVELOPMENTS

A development program is continuing which is directed toward achieving the same beam performance with a gridded electron gun with high amplification factor. A negative grid design is being pursued which uses a coarse grid of robust dimensions that is completely nonintercepting. Figure 7 shows the measured grid characteristics for this design. A grid pulse voltage of 500 V would be adequate to switch between sync and blanking levels. This grid design preserves the nonintercepting feature of the BCE while allowing increased amplification factor. Higher values of amplification factor can be achieved by using a finer grid mesh. This first design uses a very coarse mesh of only seven cells but represents a very simple and robust design.

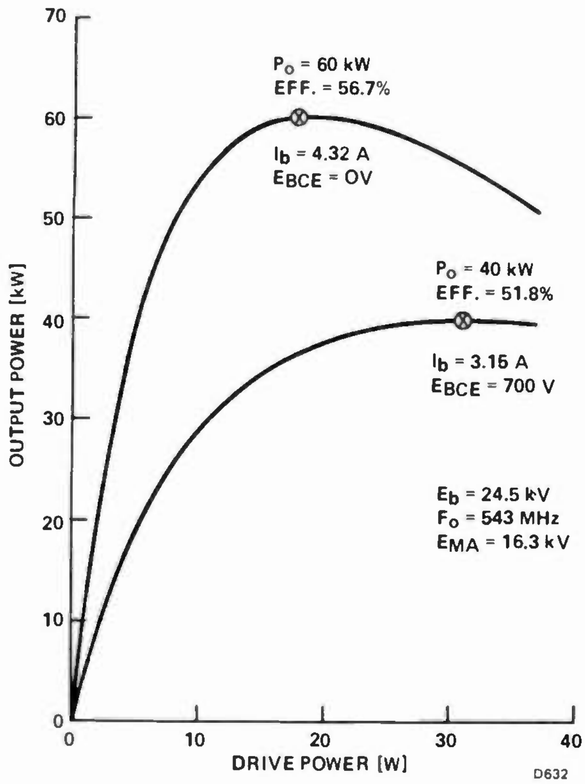


FIGURE 5. KLYSTRON PERFORMANCE

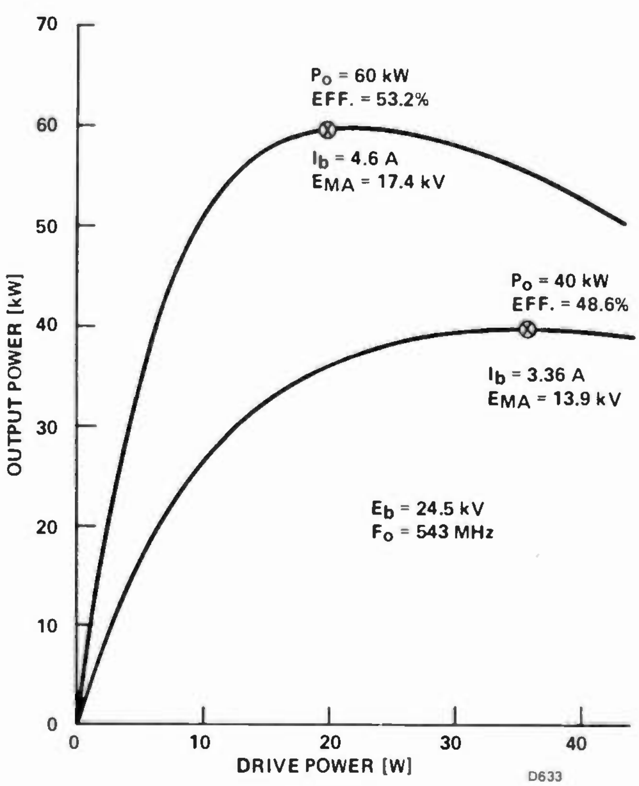


FIGURE 6. PERFORMANCE OF STANDARD KLYSTRON

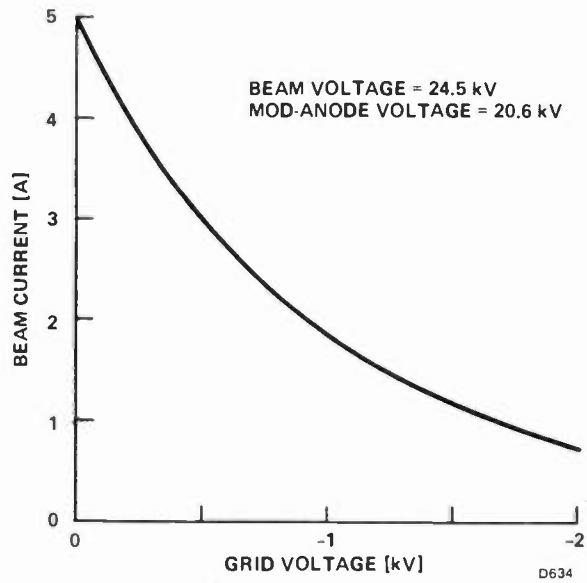


FIGURE 7. GRID CONTROL CHARACTERISTICS

CONCLUSIONS

Continuing development efforts have resulted in improvements to the Varian S-Tube design. The S-Tube uses a five-cavity design which already provides the highest efficiency currently available. The five-cavity design enhances efficiency typically by 18% over four-cavity designs (saturation efficiency of 52% compared with 44%). The new design, called the SE, provides even improved rf performance while incorporating a beam control electrode to allow beam modulation with reduced pulse voltage. The new design is being implemented in the product line as part of Varian's continuing effort to provide the most modern technology. The design uses a minimum of modifications so that continued reliable performance can be assured. In addition, the new design is provided with no increase in price.

Beam pulsing allows improvement in transmitter efficiency such that the final overall efficiency is proportional to the basic tube efficiency (2). Consequently, it is important to maximize the klystron efficiency to achieve the full benefits of beam pulsing. The SE design has accomplished that objective and provides the highest efficiency presently available for TV service. It will therefore allow transmitter operation with minimum operating cost.

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1. H. Foster, "Recent Developments in High Efficiency Klystrons" NAB Convention, Las Vegas, April 1984.

2. N. Ostroff, A Whiteside, L. F. Howard, "An Integrated Exciter/Pulser System for Ultra High Efficiency Operation", NAB Convention, Las Vegas, April 1985.

CIRCULAR WAVEGUIDE FOR UHF-TV— OPERATIONAL AND FIELD EXPERIENCE

Geza Dienes
Andrew Corporation
Upland, California

Circular waveguide is an attractive transmission line for high power UHF television. This paper is intended to describe actual performance and field experience with Andrew circular waveguide installations. To fully understand the explanation, a brief description of the advantages of circular waveguide follows.

Circular waveguide should be considered as a transmission line whenever one or more of the following conditions exist.

1. The transmitter power is 110KW or more.
2. The transmission line run is long.
3. Attenuation of the transmission line must be kept to a minimum.
4. Very high reliability is required.
5. Wind loading on tower is a cost or structural factor.

THE ADVANTAGES OF CIRCULAR WAVEGUIDE

Circular waveguide, as all other waveguides, is basically a hollow metallic tube. Whereas the cross section for rectangular waveguide is a rectangle with a roughly 2:1 ratio of top to side wall, the cross section of circular waveguide is circular. Waveguides, not having a center conductor, develop all heat losses on the surface of the hollow tube. This surface is quite large and in direct contact with the environment so the waveguide outer surface is cooled very well. In addition, the attenuation of waveguide is lower than the attenuation of coax which means that less RF energy is turned into heat in the waveguide system. These two factors combine to produce a transmission line with very high power handling capabilities. Circular waveguide has an attenuation that is even lower than rectangular waveguide and is consequently capable of handling higher powers. The power handling capability of waveguides for UHF-TV is in the range of 1 Megawatt or so, which is far greater than the output of even the highest power transmitters available today.

Because circular waveguide is a round hollow tube, pressurizing the waveguide does not deform its shape like it does for rectangular waveguide. The uniform cross-section of circular waveguide, which is retained even under pressure, produces a very low reflection from the waveguide segments and their connections. In fact circular waveguide is essentially transparent to RF energy.

Rectangular waveguide will deform under internal pressure creating periodic discontinuities along the transmission line. This could result in high VSWR and ghosting. Circular waveguide is not subject to this problem.

Circular waveguide will produce a windload which is only 1/2 to 2/3 of the windload produced by rectangular waveguide. This means that the windloading on the tower is reduced if circular waveguide is employed as a transmission line as opposed to rectangular waveguide. The wind pressure acting upon a circular waveguide run is independent of the direction of the wind.

PRODUCT DESCRIPTION

The Andrew circular waveguide product line consists of flanged straight sections of 12' long circular waveguide, a vertical hanger system to support the waveguide run on the tower, horizontal hanger system to support the waveguide run from the tower to the transmitter building, a mode filter to suppress unwanted trapped energy in the waveguide, input and output circular to rectangular transitions, and an axial ratio compensator. Various other ancillary components are also included as depicted in Figure 1.

I. ACHIEVABLE PERFORMANCE

VSWR

Circular waveguide does not exhibit band reject VSWR spikes associated with coaxial and rectangular waveguide transmission lines because of the extremely small mis-match at the circular flange junctions. The VSWR performance of a circular waveguide system therefore is basically limited by the VSWR performance of devices attached at each end of the waveguide run. These devices are the in and output rectangular to circular waveguide transitions, and the mode filtering devices. The input devices accomplish interfacing with the transmitter while the output components provide interfacing with the antenna as well as the necessary filtering. In practice these components may be designed and tuned to provide a VSWR on the order of 1.02-1.03

ACW WAVEGUIDE SYSTEMS

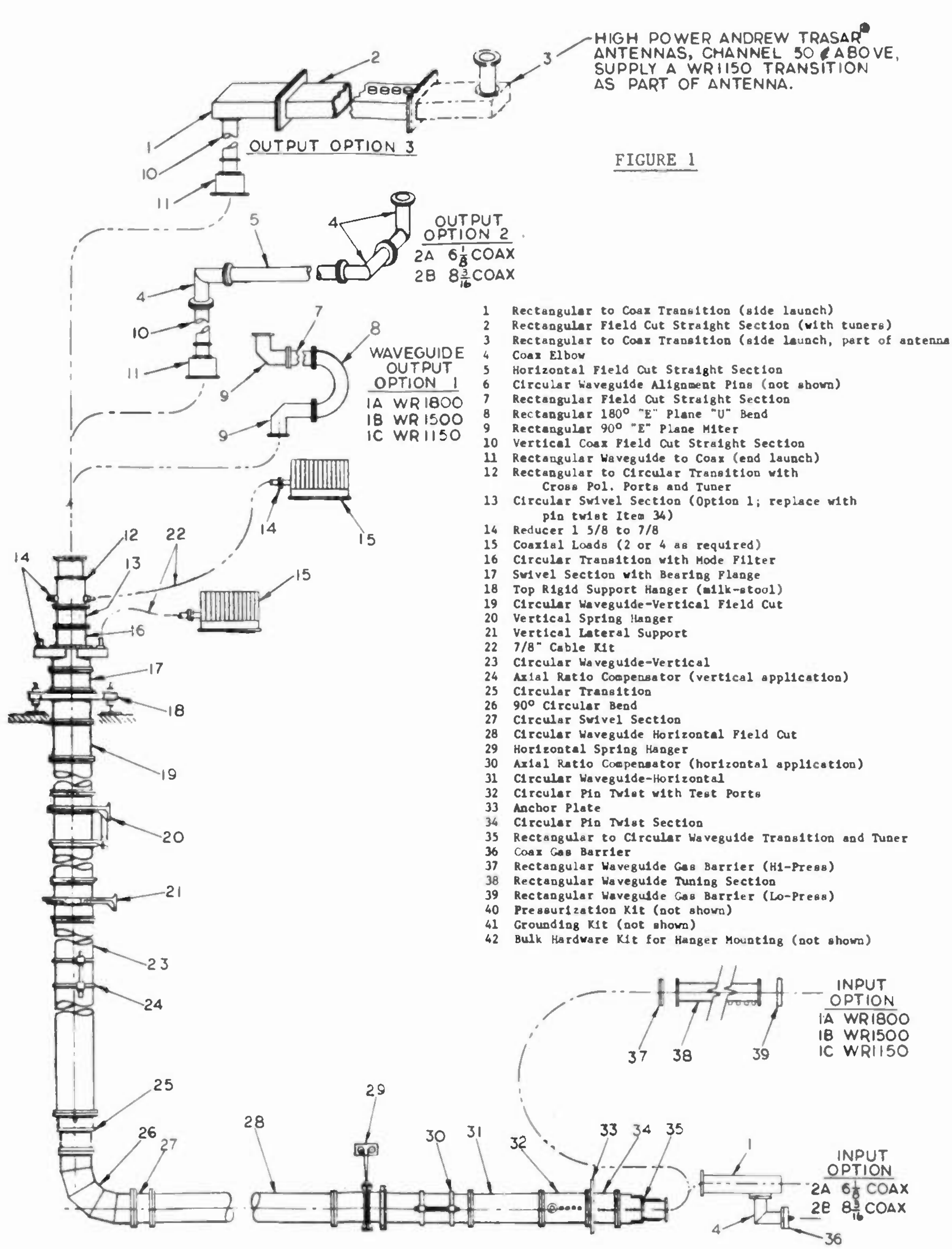

FIGURE 1

which combines with an essentially perfect waveguide run and will add up to a maximum of 1.08 VSWR. The Andrew specification for a circular waveguide system is a maximum VSWR of 1.08 or better over a channel with optimization to 1.05 or better at visual carrier if possible. RF pulse measurement techniques can be implemented to insure absolute minimum VSWR from the top end which is most relevant when considering ghosting.

RECONVERTED MODE LEVEL

Reconverted mode level is a measure of that portion of trapped energy in the waveguide that after a time delay is able to escape and be re-radiated by the antenna. With the proper filtering it is possible to supress the level of the reconverted energy to negligible levels. The Andrew specification is -50dB or better for all possible trapped mode at the system output. Considering that the effect of a -50dB reconverted mode level is approximately the same as the effect of a 1.006 VSWR it is readily apparent to be so small as to be negligible.

PRESSURIZATION

Circular waveguide should be protected against the entry of moisture with internal pressurization. The waveguide is capabile of maintaining pressures in the range of 1-2psig. This is sufficient to prohibit the entry of moisture into the waveguide system and at the same time large enough to prevent pressure drops into the negative pressure range, even if a sudden temperature change occurs.

II. SUMMARY OF FIELD EXPERIENCES

To date Andrew has supplied over 23,000 feet of circular waveguide. A total of 16 circular waveguide installations were provided. 15 of these installations also utilized an Andrew TRASAR UHF television broadcast antenna. One of the installations used an antenna from a different manufacturer. TABLE I illustrates the system parameters for these 16 installations.

TABLE I

ANDREW CIRCULAR WAVEGUIDE SYSTEMS IN OPERATION

STATION	LOCATION	CHANNEL	LENGTH	SYSTEM INSTALLED	MODEL	POWER (KW)	ANTENNA
WNYB	Buffalo, NY	49	1075'	5/87*	ACW1500	120	TRASAR**
WPTF-TV	Durham, NC	28	2029'	12/86	ACW1750	240	Other
WLFL-TV	Raleigh, NC	22	1650'	10/86	ACW1700	240	TRASAR**
KDTX-TV	Dallas, TX	58	1405'	12/86	ACW1350	120	TRASAR**
WJZY-TV	Belmont, NC	46	2025'	10/86	ACW1500	220	TRASAR**
KMJD	Pine Bluff, AR	38	2019'	3/86	ACW1750	116	TRASAR**
WDKY-TV	Lexington, KY	56	1140'	1/86	ACW1350	120	TRASAR
KRRT	Kerrville, TX	35	1575'	11/85	ACW1750	118	TRASAR**
WGBS-TV	Philadelphia, PA	57	1100'	10/85	ACW1350	120	TRASAR**
KTHT-TV	Houston, TX	67	1880'	11/85	ACW1350	110	TRASAR
KCPM	Chico, CA	24	1000'	8/85	ACW1750	60	TRASAR**
KMBH	Harlingen, TX	60	1300'	7/85	ACW1350	59	TRASAR
KTZZ-TV	Seattle, WA	22	800'	5/85	ACW1700	80	TRASAR**
KHTV	Houston, TX	39	2100'	3/85	ACW1500	110	TRASAR
KTKA-TV	Topeka, KS	49	1500'	11/84	ACW1500	55	TRASAR
KIHS-TV	Los Angeles, CA	46	300'	3/84	ACW1500	110	TRASAR**

* Planned installation date
** Circularly or elliptically polarized antenna

CROSS POLARIZATION LOSS

Circular waveguide is able to propagate energy in two mutually perpendicular polarizations. In a typical circular waveguide run for UHF television, only one of the polarizations is used. Any energy that may occur in the other polarization is unwanted and must be disposed of. The level of energy existing in this unwanted polarization has to do with the quality of the circular waveguide segments, and factors that tend to distort the circular shape of the waveguide. The proper waveguide hanging system is essential in order to minimize energy in the wrong polarization. In a properly designed and installed circular waveguide system the amount of energy in the unwanted polarization is less than 1% under normal operating conditions and less than 2% during extreme environmental conditions. To prevent even this low level of energy from interfering with the primary signal, a filter is provided which will absorb any RF energy in the wrong polarization. The loss due to this filtering is 1% of the input power by specification but is much less in practice.

POWER HANDLING CAPABILITY

All Andrew circular waveguide systems are designed to handle a power input of 240KW minimum.

Although circular waveguide is capable of carrying enormous power levels a limitation is imposed on the transmission line system by the components connected to it. Transitions to coaxial lines will naturally be associated with the power rating of the coaxial components. The power handling capability of any transmission line system is limited by the component with the lowest power rating.

TABLE II summarizes the measured system VSWR, cross polarization level and reconverted mode levels for all installations. All values are worst case results across the 6 MHz channel. The VSWR and reconverted mode level measurements were performed on the combined waveguide/antenna system. Measurements were taken at or as near to the diplexer output as possible so that the complete system (including elbow complexes etc.) with the antenna was being tested.

From this data it is apparent that the length of the circular waveguide run is not an influencing factor on VSWR performance. This of course is expected as circular waveguide is essentially transparent and exhibits extremely low VSWR.

THE EFFECT OF THE ENVIRONMENT

Many potential users have expressed a great deal of concern regarding the electrical stability of circular waveguide systems. Early circular waveguide installations (not supplied by Andrew) exhibited a disturbing tendency toward unstable performance. Both VSWR, mode conversion, and cross polarization levels varied as functions of time, and the outside environment. On some installations VSWR and apparent mode conversion levels changed from hour to hour, day to night and appeared to be highly dependent on the weather. Severe ghosting and unacceptably high VSWR occurred now and again.

In general picture quality was degraded during those times where the waveguide exhibited high VSWR or high mode conversion levels.

With this in mind all Andrew circular waveguide installations were carefully monitored for any signs of unstable performance. None of the installations exhibit any tendency to degrade with time. There is always some variation of electrical parameters in any transmission line system, including circular waveguide. The important criteria is that the changes do not degrade beyond specified levels.

MEASURED VSWR

The VSWR measured on fifteen installations (one installation has not yet been final tested) indicates that the system performance, including all components such as the antenna, the waveguide run, elbow complex, all input and output devices, filters etc. is in the order of 1.1 maximum. TABLE II lists the maximum VSWR values measured over one channel. At visual carrier VSWR's on the order of 1.03-1.05 were achieved. These results compare very favorably with other antenna/transmission line installations utilizing coax or rectangular waveguide. Because the VSWR at visual carrier is much lower than those values listed in Table 2 an operator monitoring the reflected power meter on the visual transmitter would see a lower VSWR.

CROSS POLARIZED MODE LEVEL

All systems installed and tested to date exhibited cross polarized level of better than -26 dB. All but one installation had a cross polarized level of -30 dB or better. A cross polarized level of -26 dB corresponds to a power loss of 0.25% and a level of -30 dB corresponds to a power loss of 0.1%. In terms of system performance losing such small portions of the transmitter output power is negligible. It is also seen that the Andrew specification of a maximum loss of 1% was exceeded by a minimum of 4 to 1 margin and in almost all cases by a 10 to 1 or more margin.

RECONVERTED MODE LEVEL

Reconverted mode level measurements at every installation indicated levels far less than the specified -50dB. In fact at all installations the reconverted mode level was either unmeasurably small or very nearly unmeasurably small. The dynamic range of the technique used for reconverted mode level measurements is approximately 90dB, indicating that the reconverted mode levels are in the order of 90+dB below the dominant signal. Power loss and/or interference from signals of such low levels are negligible and is in fact unmeasurable.

ATTENUATION

All waveguide systems installed were tested for attenuation. The technique used for attenuation measurements forces the input signal to travel to the end of the transmission line turn around and travel back to the source. Consequently, the transmission line appears to be twice as long as it is. Since the tranmsmission line appears to be twice as long its attenuation will show up twice as large. Due to this factor the attenuation measurement is quite accurate. In all cases the measured attenuation was within measurement accuracy of the values calculated and specified. This indicates that the power loss in the filtering device is indeed negligible.

III. MECHANICAL ATTRIBUTES

SHIPPING OF SYSTEMS

Circular waveguide is shipped via surface transportation, typically by truck. The waveguide sections are stacked four-layers high on wooden supports that support them by their flanges. Loading and unloading is most conveniently done with a forklift, but manual handling is also possible as the waveguide segments are light, weighing 20 to 40 lbs. each.

INSTALLATION

The installation of circular waveguide very often coincides with either tower installation or additinal work on the tower being performed by the tower crew. The first step is to attach all hangers to the tower members. The Andrew hanger design is such that the waveguide is supported by clamping around the waveguide body rather than by hanger attachments to the waveguide flanges. This precludes the need for an expensive hanger to tower interface scheme since the hanger locations are not critical and typically the horizontal members at the tower sections can be used directly for attaching hangers. Once all waveguide hangers have been

TABLE II

SUMMARY OF PERFORMANCE MEASURED ON ANDREW CIRCULAR WAVEGUIDE SYSTEMS

STATION	LOCATION	CHANNEL	SYSTEM LENGTH	SYSTEM VSWR	CROSS POL dB	REC. MODE dB
WNYB	Buffalo, NY	49	1075'	NOT	AVAILABLE	---
WPTF-TV	Durham, NC	28	2029'	1.12	-30	-90
WLFL-TV	Raleigh, NC	22	1650'	1.13	-30	-85
KDTX-TV	Dallas, TX	58	2025'	1.09	-35	-90
WJZY-TV	Belmont, NC	46	2025'	1.10	-36	-90
KMJD	Pine Bluff, AR	38	2019'	1.14	-35	-75
WDKY-TV	Lexington, KY	56	1140'	1.09	-33	-90
KRRT	Kerrville, TX	35	1575'	1.12	-33	-90
WGBS-TV	Philadelphia, PA	57	1100'	1.08	-35	-90
KTHT-TV	Houston, TX	67	1880'	1.13	-30	-90
KCPM	Chico, CA	24	1000'	1.12	-35	-90
KMBH	Harlingen, TX	60	1300'	1.15	-30	-90
KTZZ-TV	Seattle, WA	22	800'	1.08	-26	-90
KHTV	Houston, TX	39	2100'	1.08	-36	-90
KTKA-TV	Topeka, KS	49	1500'	1.15	-35	-90
KIHS-TV	Los Angeles, CA	46	300'	1.07	-40	-90

installed attaching waveguides to the waveguide hangers could begin. The first segment is typically attached at the bottom of the vertical waveguide run and then the waveguide segments are stacked one on top of the other until the top of the tower is reached. Installation is hindered by high winds since the waveguides are difficult to handle under those conditions. Under normal conditions the stacking rate for a vertical waveguide run is on the order of 400 to 600 feet per day. During unfavorable conditions the tower crew may be occupied in installing the ice shield and the horizontal waveguide run under the ice shield.

Once the waveguide run has been installed, all flange bolts tightened, and all hanger attachments secured the end transitions are attached. At this time electrical tests concerning the waveguide system are performed. Installation of the mode filters is checked, the axial ratio compensator is installed and adjusted for optimum performance. The relationship between the waveguide and bottom and top transitions is established, so these devices may be permanently bolted in place.

After the waveguide run has been tested, aligned, and optimized, the connection between the antenna and the top end at the waveguide run is established. The inter-connection between the antenna and the circular waveguide may be either segments of coaxial line with the proper number of elbows or rectangular waveguide components. Once the installation is complete a pressure test is conducted and final system VSWR is measured and adjusted for optimum performance.

MAINTENANCE

The waveguide system does not require physical maintainance other than a normal periodic look-see examination that is normally performed on any outside installation. Pressurization however is essential. A positive pressure must be maintained in the waveguide system in order to prevent the entry of moisture. The waveguide system also serves as conduit for maintaining internal pressurization of the antenna so the pressurization actually serves to protect the entire transmission line and antenna installation.

REPAIRABILITY

Repair of a waveguide system may be needed when a component sustains catastrophic damage. If a section of circular waveguide is damaged to the point where system performance is impaired the waveguide section must be replaced. This may be accomplished by simply removing the bolts from both flanges of the waveguide section and extracting the damaged piece. The waveguide hanger system will retain the top half of the disconnected waveguide run in its confirmed position. The bottom half of the waveguide run is supported by come-alongs in both the up and down directions. The use of come-alongs serve as safety supports and as vertical positioning devices to lower and position the bottom waveguide run.

This allows easy replacement of a damaged waveguide section. In some cases the damage is not catastrophic to the point of seriously imparing operations but is of a nature that need attention. This might be the case of a bullet hole in the waveguide run. The bullet hole will somewhat impair operations in that it will allow pressure to escape from the waveguide system and may cause some VSWR degradation as well. To repair damage like this one would need to remove all burrs or metalic protrusions that extend into the waveguide and cover the bullet hole with a sealant or with a strip of thin aluminum sheeting held onto the waveguide body with two large hose clamps.

The waveguide is fairly resistant to damage because large round tubes have very good structural integrity. We have never seen damage to the waveguide caused by handling, shipping or during installation.

CONCLUSION

A complete circular waveguide product line is offered to the UHF Television Broadcaster. Sixteen systems were delivered to date of which fifteen are in operation. All operating systems have been carefully scrutinized and exhibit stable operation with VSWR, attenuation, and reconverted more levels consistent with the requirements of high quality broadcasting. The first Andrew circular waveguide installation has been in operation for three years. The systems in operation represent an experience of over eighteen installation years. During this time many stations experienced severe weather changes. The waveguide runs were subjected to a great number of temperature cycles. To date no field problems have been found in these state-of-the-art UHF circular waveguide systems.

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UHF-TV KLYSTRON MULTISTAGE DEPRESSED COLLECTOR PROGRAM THIRD REPORT

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1. INTRODUCTION

This program involving the incorporation of depressed collector technology into UHF-TV klystrons was initiated in June 1984. The technology is expected to significantly improve transmitter efficiency. The development program is supported by a cooperative group including NASA, NAB, PBS, transmitter manufacturers and Varian. The goals and objectives as well as preliminary design considerations were presented at the 1985 NAB Engineering Conference (1). During the following year a final design was accomplished which was described at the 1986 NAB Conference (2). This past year we have proceeded to construct an experimental model using a 60 kW VKP-7555S klystron as a test vehicle.

2. PROGRAM SCHEDULE

Figure 1 shows the overall program schedule. Design Tasks 1 and 2 are completed, resulting in a design which should meet all the program objectives. Task 3 addresses the development of coating techniques for providing a low secondary yield for the collector electrodes. A facility has been established capable of sputter coating carbon onto the large sized electrodes used in this design. Tasks 4 and 6 address construction of the experimental model. Construction difficulties have delayed completion of this part of the program. These difficulties have been overcome and we are ready to proceed with Task 7, evaluation of the experimental klystron, using the test facility developed in Task 5.

3. PROGRAM ACCOMPLISHMENTS

The collector design which is being implemented is shown in Figure 2. The collector is composed of five electrodes mounted between ceramic rings to provide electrical insulation while completing the vacuum envelope. The electrodes are carbon coated in the regions of electron impingement to minimize the adverse effects of secondary emission. The sputter coating facility shown in Figure 3 was established to allow coating of the electrodes.

The collector has been fabricated along with a VKP-7555S 60 kW UHF-TV klystron which acts as the test vehicle. A photograph of the assembled collector is shown in Figure 4. A construction problem was encountered which resulted in breakage of the small ceramic ring between the klystron and the first electrode. Evaluation of the problem resulted in a simple solution, but this process caused a significant program delay.

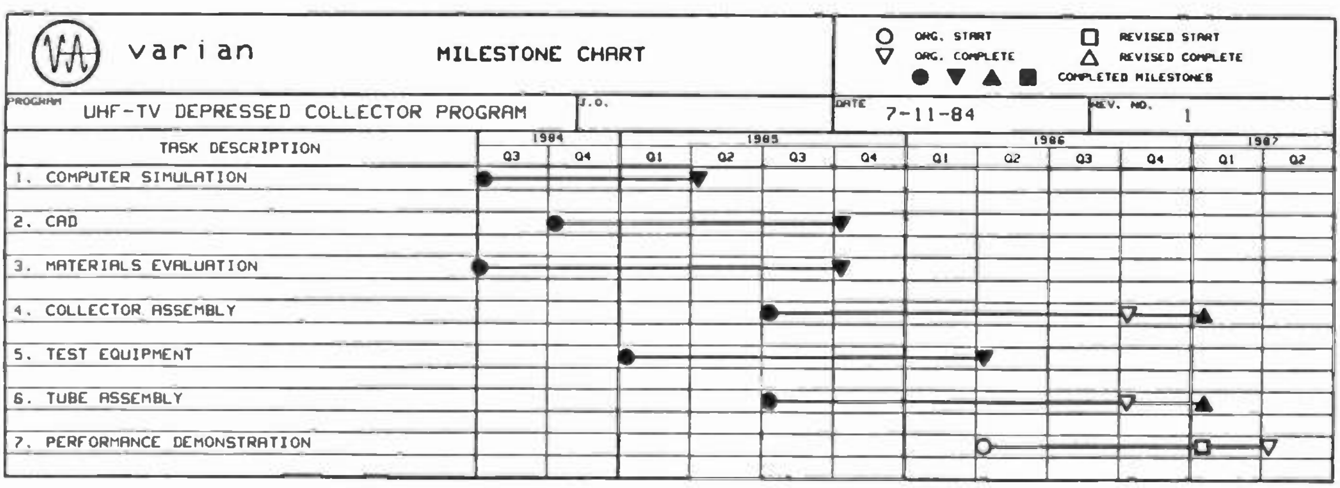

FIGURE 1. PROGRAM SCHEDULE

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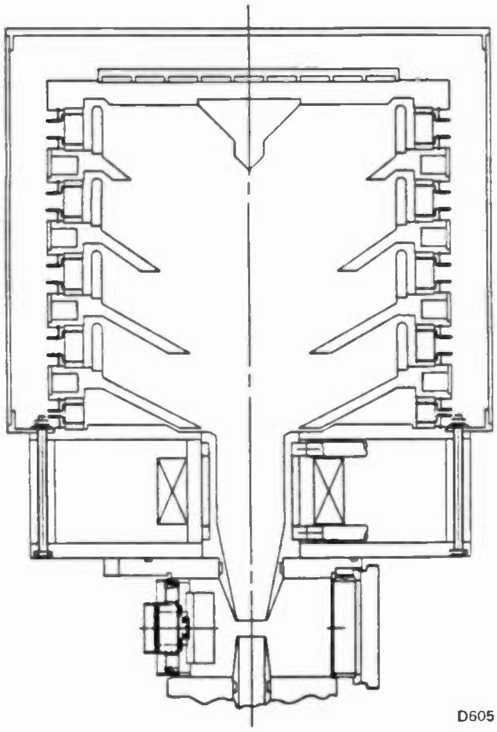

FIGURE 2. COLLECTOR DESIGN

FIGURE 3. COLLECTOR ELECTRODE COATING EQUIPMENT

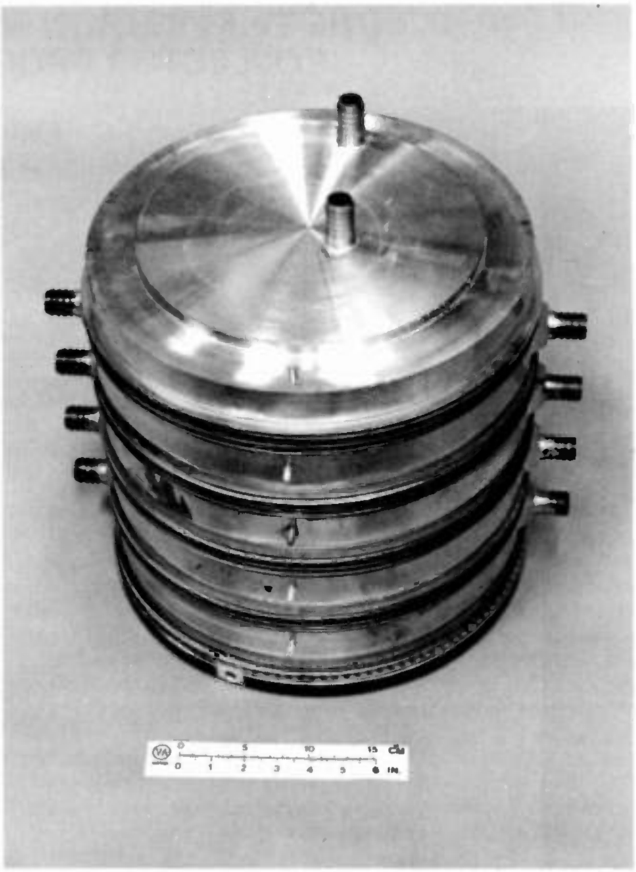

FIGURE 4. ASSEMBLED COLLECTOR

4. PROGRAM SCHEDULE

The construction of the experimental MSDC klystron is completed and it is currently being processed. Test and evaluation will follow directly. We anticipate that the preliminary performance data will be available for presentation at the conference.

5. REFERENCES

1. E. W. McCune, "UHF-TV Klystron Multistage Depressed Collector Program - A Progress Report", NAB Engineering Conference, Las Vegas, April 1985.

2. E. W. McCune, "UHF-TV Klystron Multistage Depressed Collector Program - Second Report", NAB Engineering Conference, Dallas, April 1986.

SUBJECTIVE TESTS ON A SWITCHLESS COMBINER TV TRANSMITTER SYSTEM

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ABSTRACT

The requirements for isolation in TV transmitter switchless combiners will be identified according to relevant performance parameters. The concept of testing one transmitter with test waveforms while the other transmitter is on the air will be analyzed and field results will be presented.

INTRODUCTION

A relatively new system being offered by television transmission equipment manufacturers is a high power transmitter combiner that offers the broadcaster a significant advantage. When one transmitter of a dual transmitter system fails, the remaining transmitter may be applied to the antenna at its full power without going off the air for the duration of the time it normally takes for coaxial switches to reroute the signal. Phase shifters located in the high power signal path are manipulated so that one transmitter is routed to the antenna while the other transmitter is routed to an appropriate dummy load. The generic term applied to this type of combiner system is "switchless combiner".

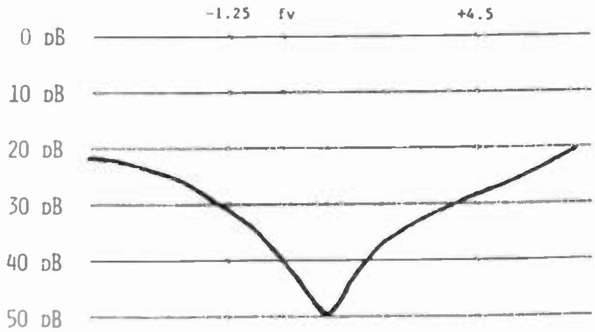
Let's take a typical example of how this would occur in a Harris dual transmitter. Assume that the dual transmitter system is licensed for 30 kilowatts at the transmitter output and that each transmitter is capable of producing a full 30 kilowatts. Obviously, at this point each transmitter is developing only 15 kilowatts. Should one transmitter fail, for a short period of time the combined output would only be 7.5 kilowatts. The phase shifters are energized and then the system is producing 15 kilowatts. The transmitter is then automatically boosted to 30 kilowatts producing full system power. This takes place in the matter of a few seconds.

Another advantage to this system is the capability for evaluation and setup of the failed transmitter while the other transmitter is on the air. Once the combiner system has energized the proper phase shifters for the correct operating mode, the failed transmitter about to be rejuvenated may be energized in an attempt to bring it back to operational quality.

Assume that the fault source has been identified, fixed, and the transmitter is now producing RF power. In an ideal system, both transmitters are producing the desired RF power and performance and there is no influence on either system's performance by the other. Now let's talk about the real world. Unless there is perfect isolation in the hybrids used in this switchless combiner, there will be some amount of crosstalk from one signal path into the other through this high power combiner. Thus, On-Air picture quality is dependent on the isolation inherent in the switchless combiner.

Our tests show that the maximum achievable isolation in the hybrids of the RF combiner system is not sufficient to prevent crosstalk which will be observable on a demodulated TV receiver or monitor. The type of crosstalk observed will depend on several factors and the most applicable ones will be addressed in this paper.

It is the intent of this paper to show how this crosstalk takes place, indicate typical values of the crosstalk, and include some subjective displays of measured values to establish a baseline for acceptable performance. Then we shall address means to reduce the crosstalk to a negligible value so that a TV station's chief engineer will feel very comfortable with the concept and practice of using switchless combiners.



3 dB Hybrid Isolation

FIGURE 1

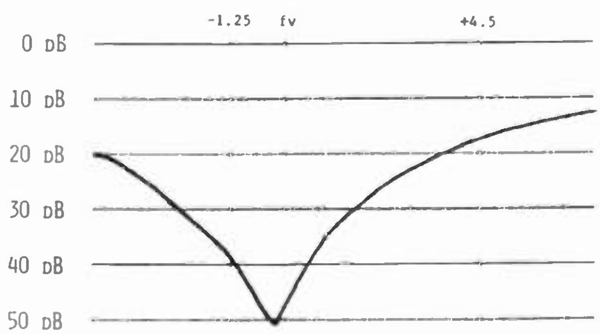


FIGURE 2

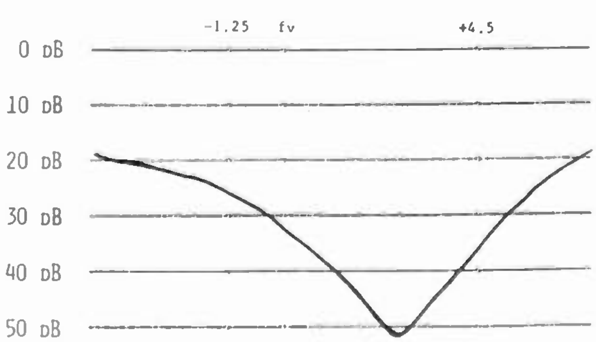


FIGURE 3

Figure 1 shows the typical isolation achieved by a 3 dB hybrid. Depending where in the channel the hybrid is optimized, the isolation could vary between that shown in Figure 2 or Figure 3. These hybrids are used to form the switchless combiner system and thus the isolation of the hybrids is the determining factor in how much crosstalk will appear in one signal path from the other.

Let us assume that the transmitter system has been energized in the "Transmitter A--Antenna and Transmitter B-- Reject Load" mode. Now let's assume that both transmitters are producing the same power output and, for the purposes of illustration a video sweep signal is applied to both. If we measured the output from the A transmitter at the antenna output port of the combiner and then measured the output from the B transmitter at the antenna output port, the display would look like Figure 4. This indicates how much isolation the switchless combiner system has.

Since the isolation is not constant for all frequencies in the visual pass band, the broadcast equipment manufacturer must be careful to specify the isolation across the channel and not necessarily at just one frequency such as visual carrier. If any compromises are to be made in the system, it is also most important to know in what frequency range the isolation is most important.

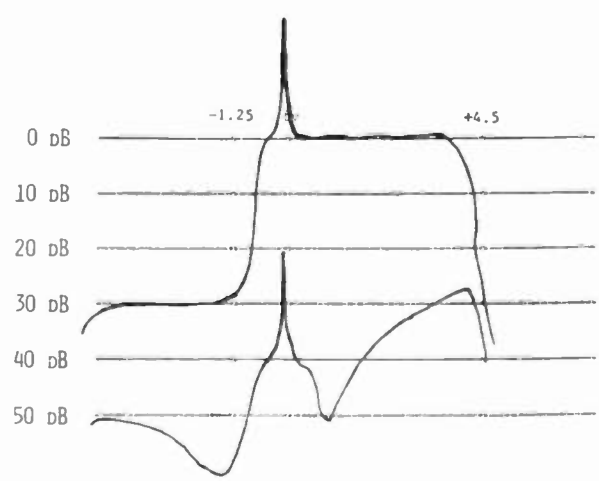


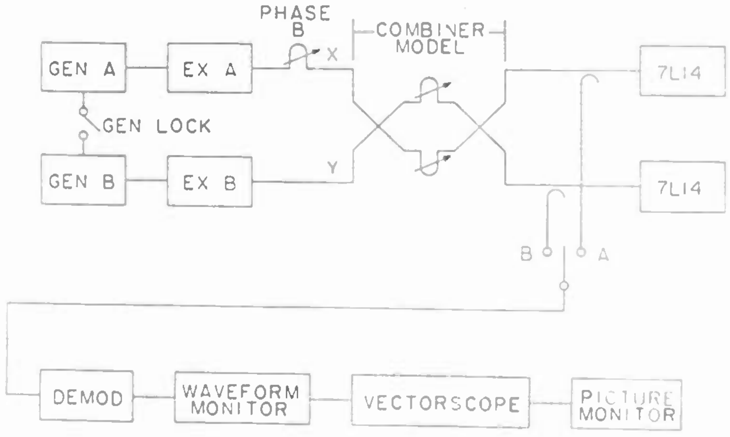
FIGURE 4

For maximum redundancy, dual transmitter systems employ two separate exciters. During the mode we have previously described, it is most likely that the A exciter will drive the A transmitter and the B exciter will drive the B transmitter. This implies that the program video would be fed to the A transmitter and test waveforms can be applied to the B transmitter. If the exciters are not of exactly the same frequency, a visible beat pattern may appear in the demodulated signal. Using the typical isolation figures of 40 dB from the switchless combiner hybrids, the beat pattern is visible and definitely objectionable.

Even though we may take great pains to adjust the frequencies of the two exciters to less than one hertz difference, we still see the crosstalk though now it will appear as a slowly alternating luminance level of the picture. If the sync sources for A and B exciters are also different, the resultant picture will have superimposed in the background the test patterns but they may be rolling or randomly moving through the picture. The effect of the random movement may be easily eliminated by the use of the same sync source for both the ON AIR signal and the test waveforms.

One question to be answered in this paper is how much crosstalk can be tolerated by the discriminating observer and how can the crosstalk be reduced to an acceptable level. In order to answer that question, various schemes were evaluated using a switchless combiner constructed as shown in the block diagram of Figure 5. System performance was evaluated for various combinations of video and RF sources. To then confirm the small signal results, full power system tests using a Harris TVD-60L with its switchless combiner were performed. These tests were performed at channel two, which is the worst case when the percentage of the channel bandwidth compared to the RF carrier frequency is considered.

We employed two video generators, A and B, feeding two standard Harris TV exciters. The video generators could be switched into a locked or unlocked mode of operation. In the locked mode, the sync pulses and color burst were coherent from each generator. Lock was verified by viewing the output on a dual trace scope. Using 3 dB hybrids and a line stretcher, a typical switchless combiner was constructed. A spectrum analyzer was coupled to each output port of the combiner to allow measurement of isolation. In addition to the isolation of the combiner system, more or less isolation could be achieved by adjusting the power of one exciter while measuring the power of each output terminal. Samples of each output were demodulated and viewed on a waveform monitor, vectorscope, and a high quality picture monitor for the final subjective analysis. Two RF visual exciters that were used could be frequency locked together or left to free-run at slightly different frequencies. By adjusting the power of each exciter, varying the frequency and phase of the RF carriers, and varying the isolation across the channel several interesting factors were learned.



MODEL SYSTEM

Figure 5

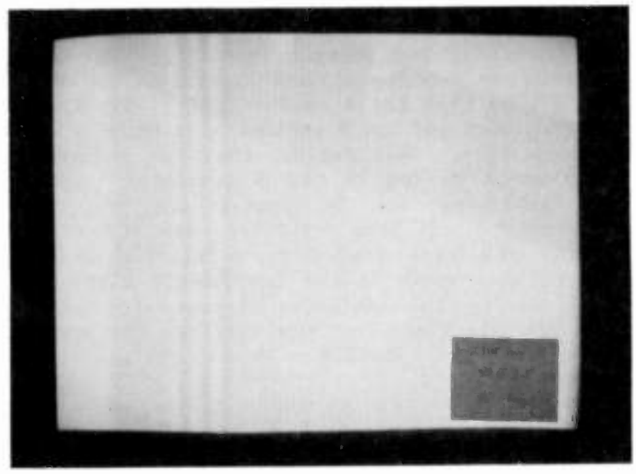
One of the most important of these factors was that the isolation was most critical in the luminance region of the visual passband. An isolation of 24 dB at color subcarrier produced no visible crosstalk in our tests.

The initial test conditions were with the two exciters at slightly different frequencies and video sources which did not use the same sync source. The resultant crosstalk was in the form of diagonal bars in the picture at the antenna port of the combiner. The diagonal bars introduced by different RF carrier frequencies could be eliminated by making the carrier frequencies phase coherent.

After this improvement was made, the crosstalk waveforms randomly jumped their way across the normal demodulated picture causing vertical bars, varying luminance levels and also test waveforms superimposed upon the demodulated picture. By using the same sync and color burst source the vertical bars were eliminated and the picture crosstalk appeared to have improved. The resultant was a stationary test waveform superimposed on the program video.

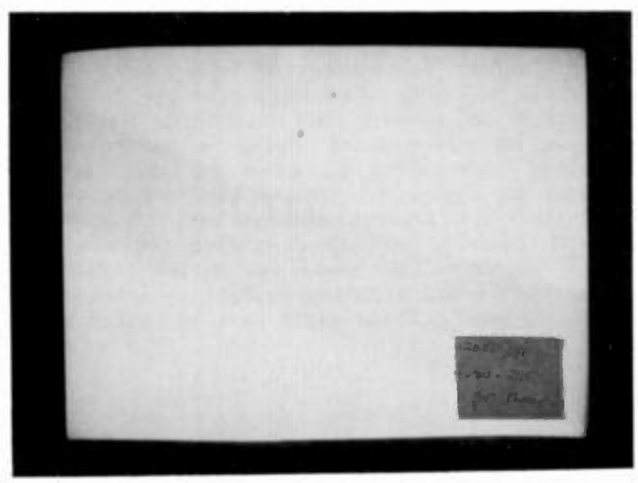
The following photographs were taken during tests on a Harris TVD-60L Harris transmitter using a high quality picture monitor. Isolation at luminance (fv + 1.5 MHz) was varied from 30 to 50 dB in steps of 5 dB. Since the isolation is not constant for all frequencies in the visual passband, the isolation measured at the visual carrier was also recorded on each picture. In all cases, one video generator was set to product a gray picture and the other video generator had either a multiburst or window pattern. The multiburst test pattern was the most visible when crosstalk was apparent.

Figure 6 represents the baseline test case of 30 dB isolation measured at the high power RF system at 1.5 MHz and 23 dB at the visual carrier with the RF carriers in phase as measured at points X and Y of Figure 5. In Figure 7, note that the multiburst pattern in the center is less pronounced as opposed to Figure 6. Here the isolation is identical to Figure 6 but the phase difference of the RF carriers is 90 degrees (phase quadrature). A convenient method of analyzing this is to view the main signal and crosstalk signals as two vectors. As shown in Figure 8, one vector represents the gray picture and the other the window test pattern. When the vectors are in phase or 180 degrees out of phase, they add together or subtract to form a more pronounced window pattern. When the vectors are in quadrature (90 degrees), the pattern is less pronounced. If phase is allowed to vary, the luminance level of the superimposed window image will change from positive to negative.



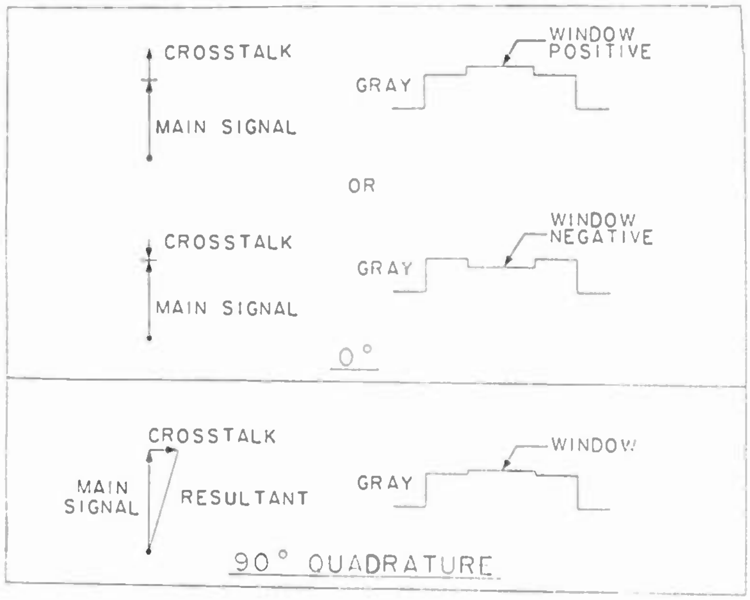
30 dB Isolation
RF Carriers In Phase

Figure 6



30 dB Isolation
RF Carriers in Phase Quadrature

Figure 7

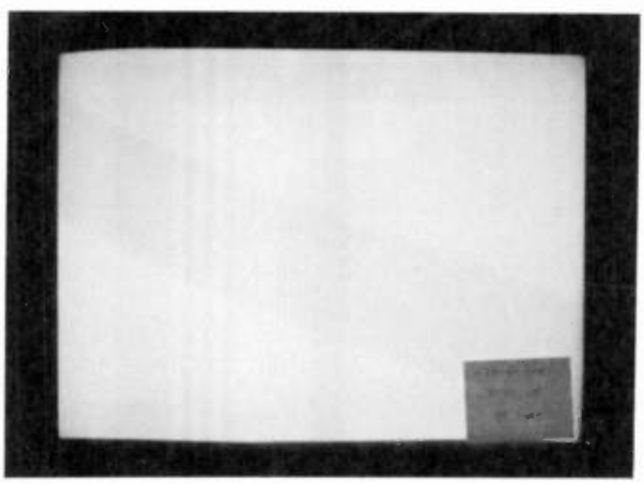


VECTOR EXPLANATION

Figure 8

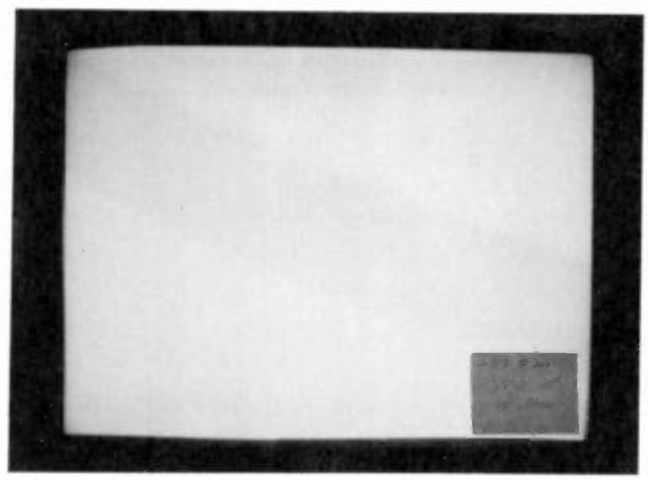
The power outputs were now appropriately adjusted so that an apparent 35 dB isolation was achieved at 1.5 MHz and about 28 dB at the visual carrier. The process of adjusting the phase difference to zero degrees was repeated with the same result except that the crosstalk was less objectionable than the previous case. This result is shown in Figure 9.

Figure 10 is a photograph of a multiburst pattern under same test condition except with a RF phase difference of 90 degrees. Note the vertical lines are less pronounced than at a zero degree phase difference. Disregard any of the diagonal bars seen on these photos, they are caused by the camera.



35 dB Isolation
RF Carriers In Phase

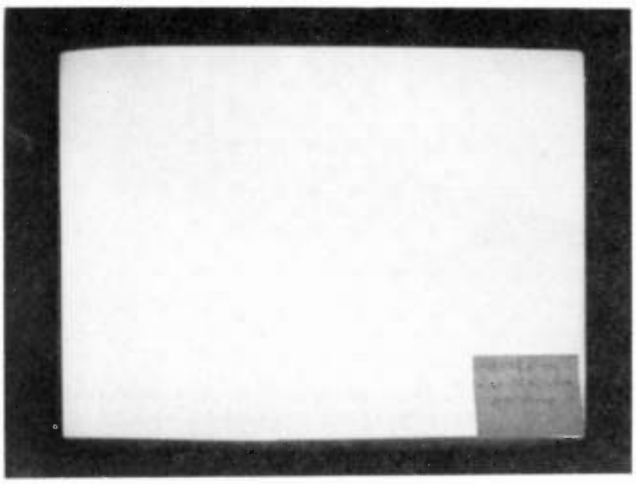
Figure 9



35 dB Isolation
RF Carriers in Phase Quadrature

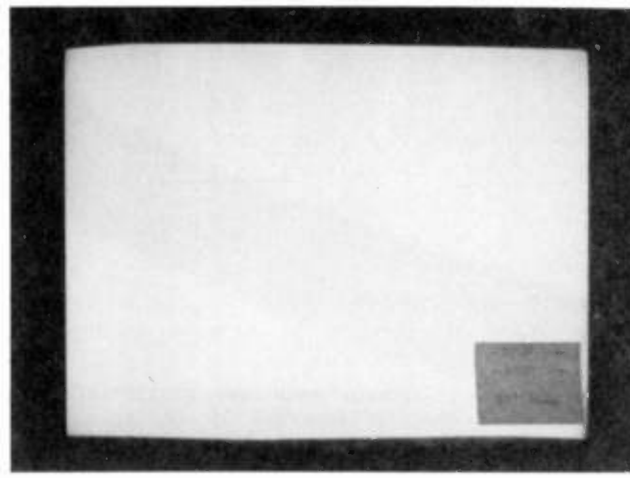
Figure 10

Let us now adjust the power outputs so that 40 dB of RF isolation at luminance is achieved. This yielded isolation of 33 dB at the visual carrier. Figure 11 shows the performance with the RF carriers in phase. Figure 12 represents the same condition but with the RF carriers in phase quadrature, which indicates a less pronounced multiburst pattern. Improving the luminance isolation again to 45 dB yields the results, as shown in Figures 13 and 14, with RF carriers in phase and in phase quadrature respectively.



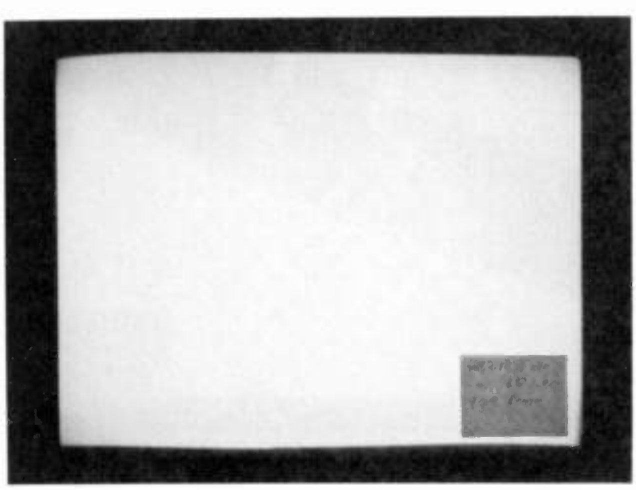
40 dB Isolation
RF Carriers In Phase

Figure 11



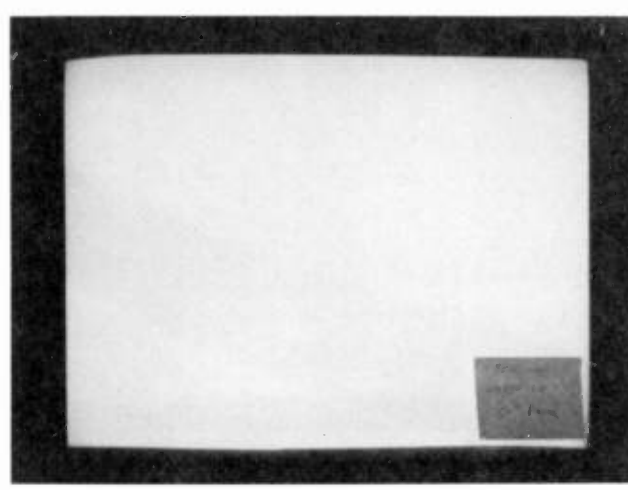
45 dB Isolation
RF Carriers In Phase

Figure 13



40 dB Isolation
RF Carriers in Phase Quadrature

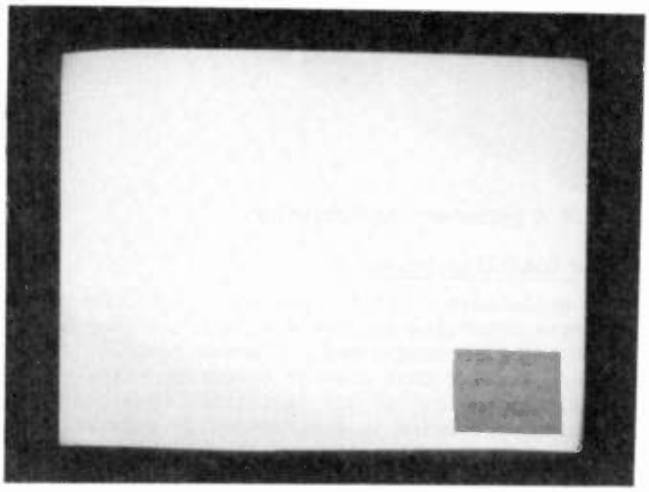
Figure 12



45 dB Isolation
RF Carriers in Phase Quadrature

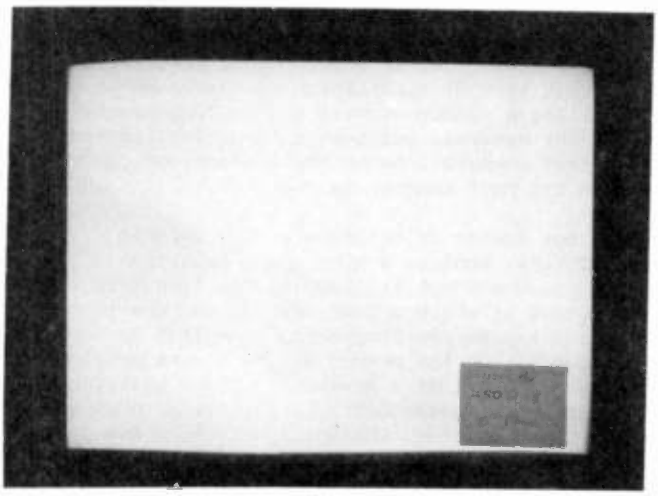
Figure 14

The final set of photographs, Figures 15 and 16, was taken with 50 dB isolation at 1.5 MHz and 43 dB at visual carrier. In Figure 15, with RF carriers in phase, the vertical lines are barely visible and when the RF carriers are adjusted for phase quadrature, Figure 16, the lines have virtually disappeared.



50 dB Isolation
RF Carriers In Phase

Figure 15



50 dB Isolation
RF Carriers in Phase Quadrature

Figure 16

SUMMARY

To take advantage of the possible features of "switchless combiners" the subject of crosstalk must be addressed and quantized. To simultaneously minimize crosstalk and achieve the ultimate performance in a switchless combiner system, four parameters must be optimized. First, the isolation inherent in the RF combiner system should be optimized. It has been found that the isolation should be optimized for the luminance portion of the visual passband. Second, the RF signals of the two exciters should be frequency coherent. Third, the RF carriers must be adjusted for a quadrature relationship at the antenna output port of the combiner system. And fourth, the video generators should be locked together.

Our tests indicate an improvement of 10 dB can be achieved by locking the video generators together and using frequency coherent RF signals.

NEW DEVELOPMENTS IN COMPUTER CONTROLLED REMOTE CONTROL CAMERA SYSTEMS

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INTRODUCTION

New developments in computer controlled, remote camera control systems have resulted from the integration of computer processing, data communication and electro-mechanical technology. It is becoming increasingly apparent that the potential opportunities presented by such integration can be directed toward News and Presentation Studios, Outside Broadcasts, Remote Studios and Parliamentary Broadcasting. The objective of this paper is to highlight applications that have been realised, with the benefits attributed to the implementation of such systems being illustrated and supported by an indepth study of a remote satellite studio application.

POTENTIAL BENEFITS

News & Presentation Studio

It can be strongly argued that for News and Presentation programmes where, by nature, the cameras are static within the studio, it is unnecessary to have them continually manned. Therefore, cost-effective deployment of manpower would be enhanced by centralising the control of all cameras in the control room allowing control of pan, tilt, zoom and focus for all studio cameras to be controlled by a single operator.

Pre-programmed positions and camera moves enable high quality camera movement to be undertaken more smoothly and accurately than under manual control. Location of the operator in the main control room also provides more direct communication between the studio director and the camera operator.

Outside Broadcast

There are situations in Outside Broadcasting that would benefit from the ability to place broadcast cameras in locations that are either dangerous or inaccessible for a cameraman to operate. This category of applications includes sports coverage and events of international interest such as Royal Weddings or summit meetings where the cameras can be mounted in locations to provide more dramatic pictures. Control of the remote camera installation could be temporarily or permanently distributed throughout the location with a central termination point. The controlling console could be removed if necessary or form part of a permanent installation.

Remote Satellite Studio

It is anticipated that increasing use will be made of remote interview studios for regional News and Current Affairs programmes. Remote control applications for this area of operation allow centralised control of all facilities required in a small studio which is geographically separated from the main studio complex. Remote control of an unmanned satellite studio negates the need for local technical and operational staff. The implementation of remote studios should result in a significant decrease in manpower resources and optimises the deployment of skilled manpower available.

Parliamentary Broadcasting

The broadcast of Parliamentary and Legislature proceedings is normally carried out from environments in which the presence of third parties is at times considered detrimental to the activities taking place. Latest developments associated with this type of application are based on installing a number of wall and ceiling-mounted broadcast cameras, automatically controlled from a central computer, hence the presence of cameramen in the main chamber is avoided.

A typical system as utilised by the Ontario Legislature, employs a microphone selection system as the input stimulus to the computer, the output of which drives and directs the cameras. The computer is pre-programmed to select on a vision switcher the output of the camera providing the best picture of a given microphone position. The central computer controls peripheral equipment which includes Video Character Generator and Down-Stream Keyer. A typical system would allow operatorless transmission of long speeches and debates, allowing manual override at any time.

REMOTE SATELLITE STUDIO APPLICATION

For the purpose of this presentation, a Satellite Studio is defined as a studio geographically remote from the main studio complex, an example being the IBA Studio at the Houses of Parliament, Westminster, which is used for interviews with Members of Parliament and political correspondents. Interviews are conducted without the need for the interviewee to leave the confines of Parliament in order to attend the main studio centre.

Implementing such a system allows 'on demand' use of a remote studio for News, Current Affairs or Parliamentary Reports, enabling broadcasts to be made without the presence of local, technical and operational staff.

SYSTEM CONFIGURATION

Central Control Console

A basic system configuration would normally comprise a Central Control Console which would be located at the main studio centre. This would be connected via a suitable communications link to the remote studio. This Control Console provides access to all remote facilities for camera and lens positioning, iris and black level, lighting, audio and ancillary equipment control.

Figure 1 shows a console used by Tyne Tees Television in England. In this particular case the camera and lens position is manually controlled by operation of the two joysticks which control the velocity of movement of the pan, tilt, zoom and focus axes. The joysticks provide proportional control, resulting in smooth on-air movement capability. In addition a number of shot positions can be preselected and preprogrammed. Individual circuits or groups of lights can be switched on or varied in intensity. Control of selected camera functions include access to white/black balance and auto-iris. When required, microphone and recording facilities can also be accessed for control and switching.

When more extensive control of a camera is required, additional facilities include those illustrated by Figure 2.,ie. control of Red and Blue lifts and gains as well as CCU centering controls.

Control of peripheral equipment such as transmission On-Air lights, studio talkback and control of studio telephones, is possible once the backbone of a computer-controlled system is established. Supporting the central console would be a distributed microprocessing system continually monitoring the operator input and transmitting the detected input as a package of digital information. The system as described can be configured to allow the operator input to be emulated by instructions from a centrally-based computer. Once the control protocol from the computer is established the computer will automatically drive camera and lighting systems without operator intervention.

Data Transmission

Data transmission between the central and remote studio would be accomplished at speeds ranging from 1200 to 9600 bits per second. Data generated by a computer or as a result of manual selection at the Control Console, is interrogated by the microprocessor located within the central console, converted into a serial data stream and routed via a modulator/demodulator (modem) to the selected data transmission link. The formatted protocol is primarily suited for full duplex operation, especially important to ensure the speedy acknowledgment of commands returned from the remote studio. In the case of data transmission between countries or continents, the protocol would be biased towards simplex operation which would avoid the necessity to tolerate up/down delays on the return path due to the use of satellite links. Such cases require a level of redundant data being incorporated to ensure quality data is received.

Remote Studio

The control electronics unit at the remote studio is connected to the receive modem. It accepts the serial data stream, checks for valid data and locally distributes the necessary command instruction to the co-located broadcast equipment. Commands can be generated in either digital or analogue form depending upon the equipment interfaces to be driven.

Camera and Lens Control

Signals for camera and lens movement control use 12-bit words which, when decoded, drive closed loop servo systems for the pan, tilt, zoom and focus axes. Typical mounting of the camera and lens is illustrated by Figure 3. Proportional and shaped control of the joystick action allows very smooth camera movement essential for 'on-air' use.

Control signals to the audio and lighting equipment are distributed in a 'star' configuration from the control equipment, with voltage levels and signal format being dependent on the equipment used.

OPERATION

When it is necessary to utilise the studio for interview purposes, the interviewee is escorted into the studio and seated facing the camera. The operator at the main studio centre, prior to broadcasting, can pre-programme camera and lens positions likely to be required for the programme. Camera line-up may also be undertaken if necessary.

Once the interviewee is seated, lighting levels can be established and suitable microphones enabled. Studio ancillary equipment such as transmission warning lights, cue lights and telephone mute may be set as required and the interview may be conducted as though it was taking place in an adjacent studio. During transmission, any control parameter can be altered.

NETWORKING

The satellite studio operation described above was based on a single point-to-point configuration. This, however, would be the most basic of networks. The concept detailed could be extended by integration of a matrix of control consoles and remote studios.

Distributed Control

The introduction of Distributed Control would be advantageous because it allows control over a remote studio from more than one physical location. Figure 4 illustrates a typical configuration which allows all remote studio facilities to be under the selective control from three different control rooms (A, B and C). In some cases it may also be necessary to split up the various control elements further, such that microphone control can be located on a sound control desk,lighting control as part of a main lighting position and camera line-up on an engineering position.

Distributed Studios

Regional broadcasting from distributed studios would require a network of remote studios which, as illustrated by Figure 5, are under the control of control rooms A, B and C, a matrix switch allowing any studio to be connected to any location.

Standby

It is always desirable to establish a permanent standby link. However, cost-effective alternatives are available by use of dial backup. Figure 6 illustrates such a configuration. In such a case it is obviously important that loss of communication does not allow any change to the status of remote equipment.

FUTURE DEVELOPMENT

Ongoing developments in this market area are focused on the ability to physically move a camera from one location to another in a remote studio. It is believed the next stage of computer-controlled remote-control television cameras will be the introduction of track-orientated positioning of the camera. In this application, local computer equipment would be able to control the physical movement of the camera across a studio floor moving, for example, from a presentation desk to an interview set, the movement to be noiseless, hence allowing 'on-air' movement to take place.

CONCLUSION

In conclusion, it is conceived that the requirement for studios to operate in a cost-effective manner, whilst maintaining an ability to broadcast 24-hour news and current affairs programmes, demands that new strategies of control are developed. Implementation of computer control remote broadcasting equipment is providing the opportunity to realise such aims.

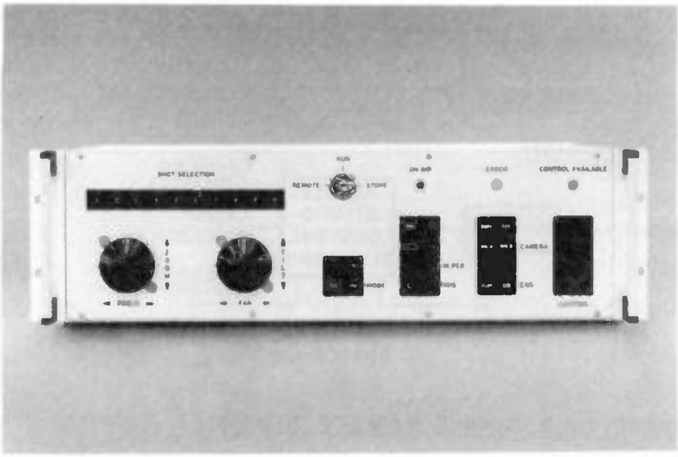

FIG. 1

TYPICAL CONTROL CONSOLE FOR
REMOTE SATELLITE STUDIO

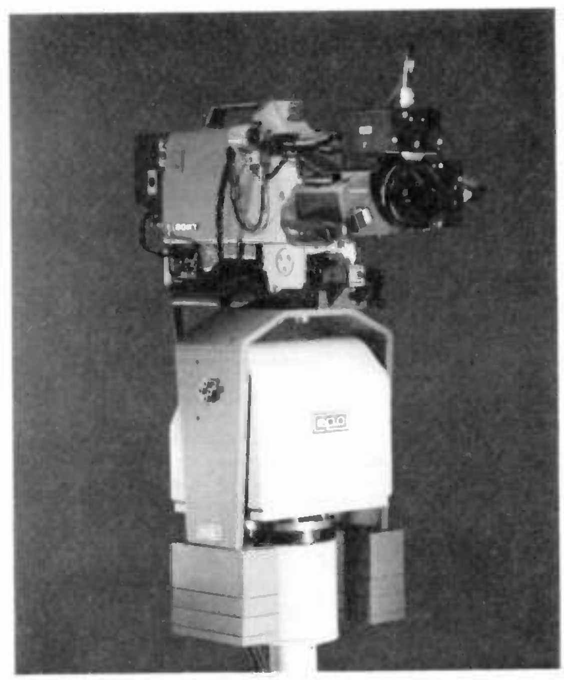

FIG. 3

EP0423 PAN & TILT HEAD AS USED FOR
TYNE TEES TELEVISION REMOTE SATELLITE STUDIO

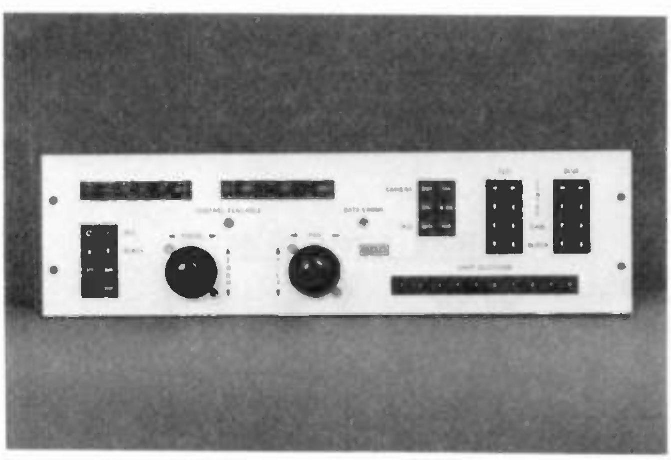

FIG. 2

REMOTE SATELLITE STUDIO CONSOLE WITH
ENHANCED CCU LINE-UP FACILITIES

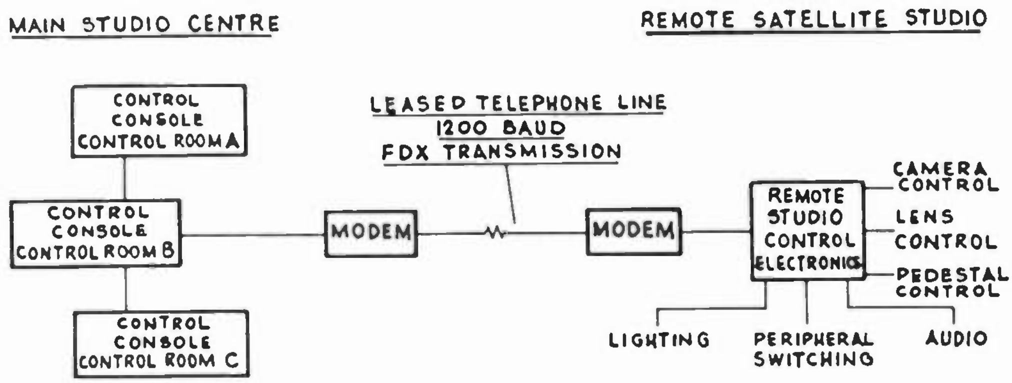

FIG. 4 DISTRIBUTED CONTROL OF A SINGLE REMOTE STUDIO

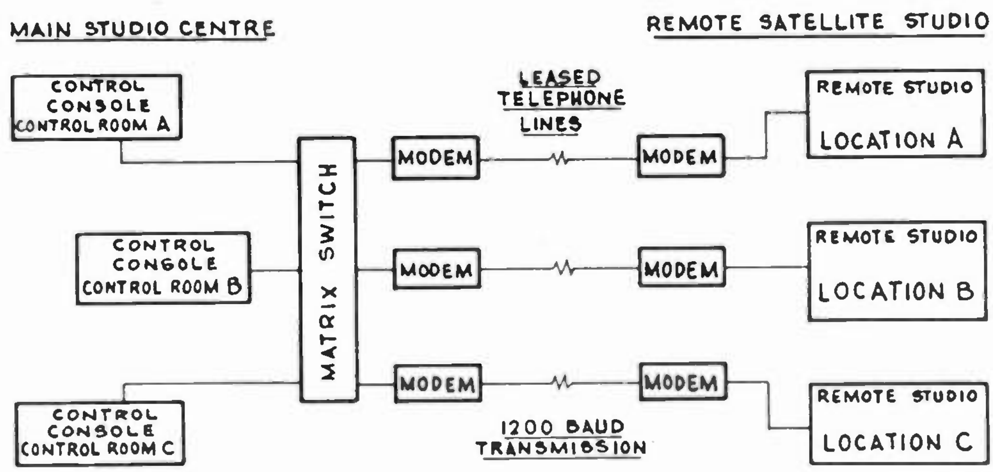

FIG.5. DISTRIBUTED NETWORK OF REMOTE STUDIOS

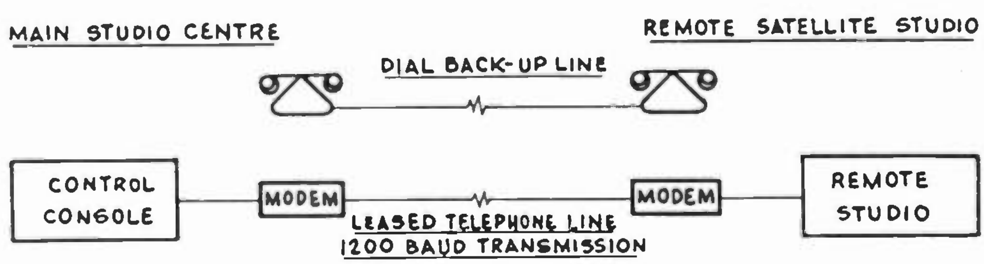

FIG. 6. DIAL BACK-UP SUPPORT FOR SINGLE REMOTE STUDIO

COMPUTER AIDED DESIGN (CAD) SIMPLIFIES AUDIO-VIDEO SYSTEM DESIGN AND DOCUMENTATION

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ABSTRACT

Computer Aided Design and Drafting (CAD) has revolutionized the design of audio and video equipment. CAD was done on mainframe and mini computers. But now CAD workstations start as low as $1000.

This session will explore the differences between CAD tools and CAD applications. Selection guidelines between CAD programs will be presented.

Sample drawings and use tips for the basic CAD applications will be presented, along with some advanced CAD applications. A complete editing suite will be designed and drawn, including floor plans, rack layouts, system block drawings, bill of materials, heat/power/weight loading, cable documentation, custom equipment designs and preparing cable labels. Several CAD users will also be available to answer specific application questions.

Computer Aided Design and Drafting (CAD) has revolutionized the design of audio and video equipment. A few of the features that make CAD helpful are:

- the precision and reproducibility of drawings
- reduction of repetitive drawing tasks
- automatic measurements and dimensioning of drawings
- electronic schematic automation

In most large organizations, CAD has been done on mainframe and mini computers. But now CAD can be done on PC's or AT's, using CAD tool programs costing from $49-$10,000; and can be printed on dot matrix printers. A basic PCXT can be purchased for under $1000 while the price of high performance PCAT's with medium resolution graphics monitors and hard disk drives has fallen under $5000, and some for under $3000. Gone are the days when you had to wait 15 minutes to see an unreadable low resolution rendition of your station floorplan. The new generation of 80386-based supermicro computers promises even greater performance. (See ATTACHMENT 1 for more hardware information.)

CAD FOR AUDIO AND VIDEO ENGINEERING

Now that it is more affordable, powerful, and easier to use, CAD also has the capability of revolutionizing station design. Areas of application include the following, and those in italics are covered here:

Basic CAD Applications

- *Architectural plans and room design*
- *Block/system drawings*
- *Cable-run planning and measurements*
- Cabinet and console design
- *Custom electronics design*
- *Ergonomic planning and simulations*
- Making PERT-type charts
- *Preparing Bill of Materials*
- *Rack and equipment layouts*
- Set planning and simulations

Advanced CAD Applications

- *Air-Conditioning specification*
- *Cable Documentation from drawings*
- *Calculating heat/power/weight loading*
- Electronic routing
- 3-D systems modeling and perspective
- Preparing cost estimates
- *Printing Cable labels*

SELECTING CAD SOFTWARE

We distinguished between purchasing a ***CAD Tool*** and a ***CAD Application***. A ***CAD Tool*** is the basic software as it comes from the publisher. Some packages have more powerful tool-sets, such as electronic symbols libraries, architectural libraries, architectural drawing enhancements, dimensioning, etc. But these are all horizontal market products. To use them, you must spend days of time building libraries of components. (It takes a minimum of 30 minutes to pull 1 spec sheet, draw a basic block component and refile the spec sheet. Our full-time artist spends nearly 2 hours per product, making 3 to 20 different perspective drawings for each product.)

A ***CAD Application*** is a tool, plus a package of plans and supplies specifically tailored to a particular industry. Our *VidCAD* and *AudCAD* libraries are applications ***specifically tailored*** for the serious audio and video engineer/designer. Of course, applications cost more, but you can turn out drawings immediately--not weeks or months later. Our customers say they have justifed the cost of each *VidCAD* library within 1 or 2 weeks. (See ATTACHMENT 2 for more software selection information.)

SYSTEM DRAWING FROM WHOSE PERSPECTIVE

During the past years of analyzing the needs of video and audio professionals in CAD, we have noted 2 major perspectives in studio design:

User/Manager/Salesman Perspective

Primarily interested in equipment features and how equipment fits into a rack or floorplan, and more concerned about *what* it will do for them or their customer than how it interfaces with the system. They will often delegate design of flow/system diagrams to engineers to interface (hopefully before it is purchased!). The User/Manager is not satisfied with drawings made up from boxes with names, but wants some detail to help imagine how it will look and feel. He is more comfortable with 8 1/2 x 11" drawings than D-size plots. (Note: The higher up in management, the more this is the case.)

Engineer's Perspective

Primarily interested in *how* the equipment interfaces into the system, and what kind of installation problems are associated with it (e.g., AC load, power consumption, delay, DA/isolation requirements, cable and connector types). The engineer is quite satisfied with (if not insistent upon) boxes bearing names to represent components, and must have large-size plots to see all the cable documentation detail that he feels is necessary.

These differing perspectives were one of the reasons we designed *VidCAD*--so that:

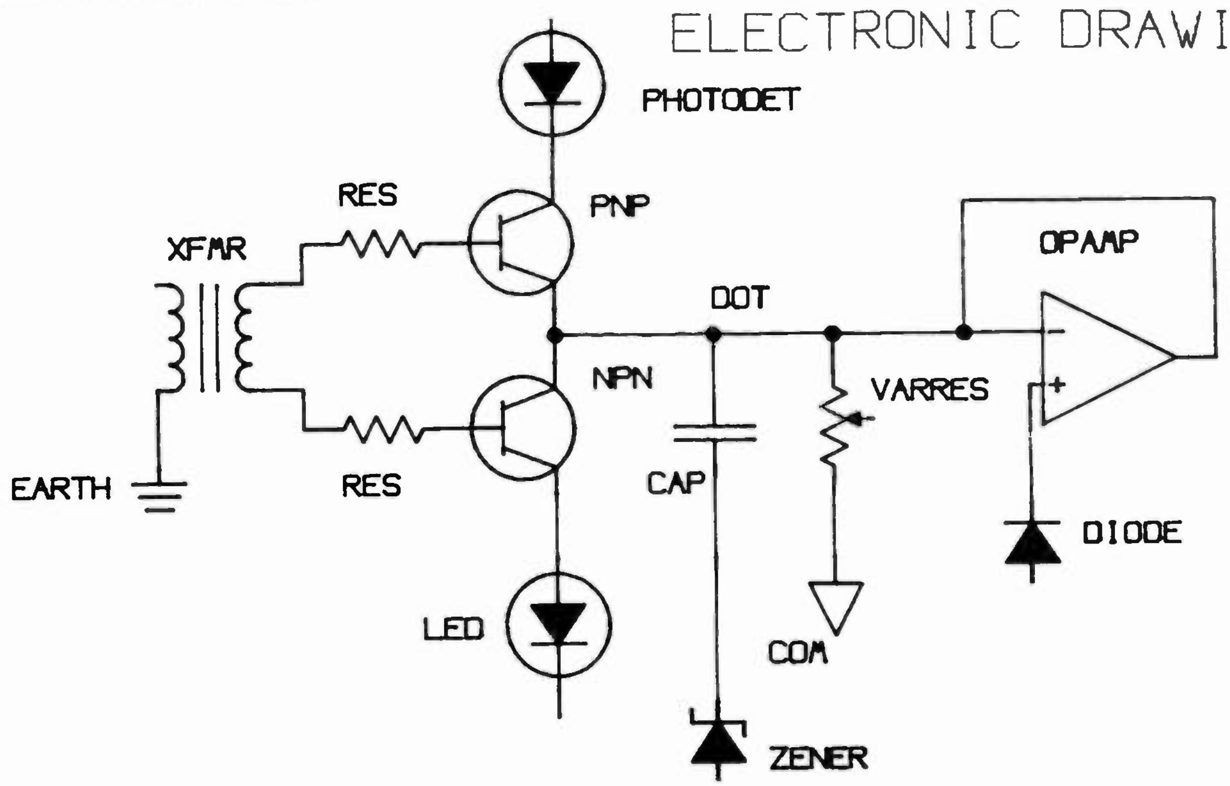
- -Little time need be spent creating a CAD drawing library of components
- -Realistic and aesthetic drawings take even less effort than blocks with names
- -Preparing Bill of Materials and Cable Documentation can be quick and easy
- -Calculating AC loads, weight and power consumption is effortless
- -Ergonomic planning is simple and available

Both of these perspectives are necessary for efficient studio design. Since the User/Manager is most often the one who gets money allocated because he has a need, the engineer who learns to use *VidCAD* as a tool to help get his desires filled, will be much happier. Some of our customers have found that 70-80% of their plots are in 8 1/2 x 11" for presentation and selling ideas to management.

In actuality, using *VidCAD* is as simple as selecting the components you want to use in the drawing, placing them appropriately, and adding system names (e.g., VTR1). This is why we call *VidCAD* an application as opposed to a tool. Most of our customers find their first drawings take a few hours, but they are useable. Later it takes only minutes, as they become familiar with the commands and program.

We will now explore several of these capabilities in detail, from the simple to the complex.

FIGURE 1:
ELECTRONIC DRAWINGS



CREATING CUSTOM EQUIPMENT

CAD allows us to easily create custom cabinets, consoles, patch bays, and electronic equipment easily. Here is an example (Figure 1) of an electronic circuit diagram prepared with our electronic symbols library. (Note: If you need complex auto circuit design and routing, one of the specialized packages will do a better job than any moderate priced CAD program. Several are available for under $1000.)

Let's create a custom patch panel of 16 1/4" jacks, 2 rows, with designation strips, 2 rack spaces high (Figure 2). Here are the steps:

a. First we make a rectangle of the correct size. CAD will allow you to do this with a mouse or by X-Y coordinates. I like to construct new components by coordinates (0,0,19,3.5 in this case). This allows precise sizing, and the 0,0 origin lets me do some easy mental math for adding other points.

b. Next, construct the first 1/4" jack mounted in a 3/8" hole at 1.5,1.

c. Copy this 16 times at 1" intervals, still using coordinates for accuracy.

d. Make the first designation strip using RECTANGLE.

e. Copy jacks and strip with window copy, then enter the text. If you use our *CableDOC* program, it is possible to extract the designation strip text for further processing (Editing, Printing) or it is possible to window copy all these designation strips from your drawing into 1 master designation strip drawing for plotting. This plot can be cut, laminated, and placed directly on the designation strips.

f. By adding dimensioning lines to the drawing, you can give the drawing to a machine shop for precision milling, or even make a life-size model and use it as a pattern. Automatic dimensioning is a must for easy custom component design, and is also helpful in architectural layouts and installation drawings, since the measurements are precise and it requires no retyping of measurements.

g. Now you may save this drawing as a component to use in other drawings. To save time and confusion, here are some tips on how we name components:

-No "-" or "," or " "(spaces) are used in names, e.g., V05850 not VO-5850. This is necessary because component names must be 8 characters maximum for storage in DOS.

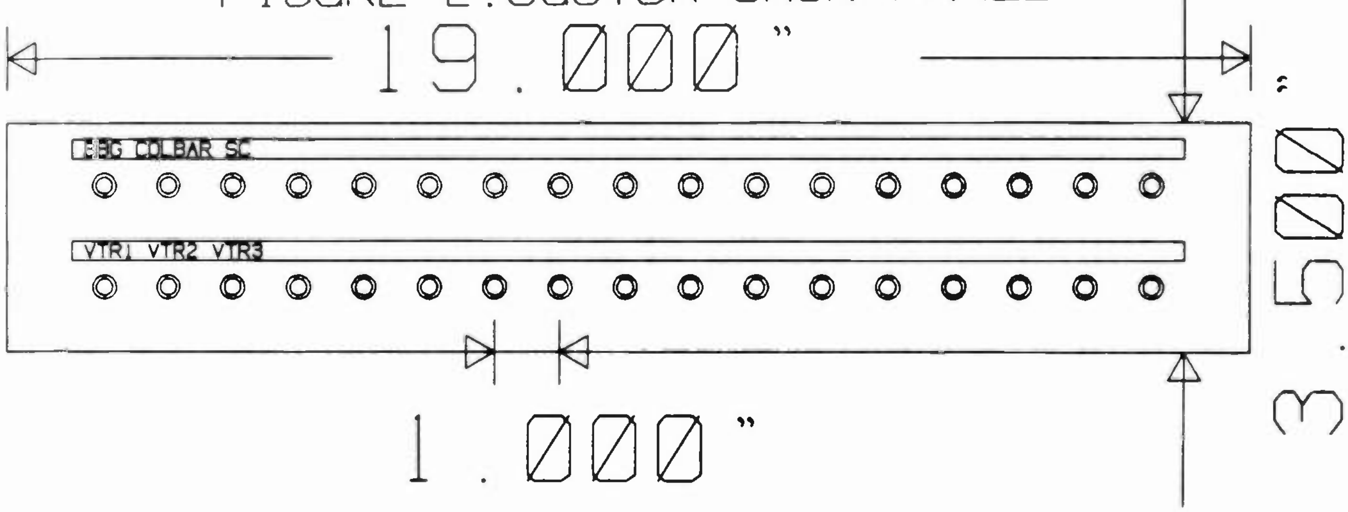
-A manufacturer's name becomes a 3-character DOS extension to the product name, e.g., Panasonic = .PAN, Sony = .SON. You must type the full name and extension (VO5850.SON) when you Component Load.

-Generic components such as tripods, racks, cabinets, and other furniture use the DOS extension .CMP which stands for component. It is not necessary to type the .CMP extension when you load it.

-For each component you entered there will be 3 or more "components" listed in sequence. The last of these is the "composite component". Do not attempt to place this one in the drawing. Use these:

B = System Block View 8003BB.GVG
R = RACK FACE VIEW 3003BR.GVG
E = RACK TOP VIEW 3003BE.GVG
F = CONSOLE FACE VIEW 3003BF.GVG
T = CONSOLE TOP VIEW 3003BT.GVG
COMPOSITE VIEW 3003B.GVG

FIGURE 2: CUSTOM JACK PANEL



-If the component normally is used with several accessory products, such as CCU's, TBC's, RCU's, KEYBOARDS, then these are normally included in the composite view, so you only need enter the one name. Some items have 10 or more views.

CREATING RACK DRAWINGS

To create rack drawings, you simply need to load the components you are using in the system from the library, and place them where you wish. This drawing (Figure 3) is being loaded from a Batch file that contains the list of components to be used in an ASCII text format. A Batch file is a simple way of loading the same components in different drawings with out re-keying. We also recommend storing drawings in this compact form because it is useful in case binary drawing file becomes corrupt. It can be used to create a Bill of Materials, *CableDOC* or *CAD Spec* document.

Reference Points--Placing components requires knowledge of the reference point. Here are our rules:

a. Most reference points are in the lower left hand corner of the component. Racks should have 1.75" rack space marks to guide your placement of components.

b. Normally studio products are rack mounted. Consequently *VidCAD* component reference points are for rack mounted versions. In other words, if the component is rackmountable, the reference point is normally at the standard lower left hand corner.

c. Since some products are also frequently used in conference room and media center applications, such as VO5800.SON, the rack ear options are not drawn, but the reference points are still as if they were rack mounted.

d. When in doubt about proper reference points, just place it in a blank spot. After all, Erase Last quickly removes any mistakes!!

Place Equipment where you like. It is a lot easier to start at the bottom of a rack and work up, but the rack markings simplify this task. If you don't like it, just move or erase it. Tips:

a. In moving and removing objects, you do not need to window the entire components but only the reference point.

b. If you have included text or other new information over the component, you must put the window around the reference point and these objects.

ERGONOMIC PLANNING

Ergonomics is the study of people as they relate to their working environment, especially the physical relationships between man and machine. Hence, in studio design, it is the study of equipment placement for ease of use and less strain. Computer simulations are one of the best ways of studying these relationships. With CAD, you can place equipment and people in the drawings in simulated scale, then check the different views (face, side, top) to approximate the suitability of these relationships. Some guidelines include:

-Eye travel + 15 -2- degrees vertical, 30 degrees horizontal.
-Normal reach from shoulder height to waist vertically.
-Working surface 26-28" in sitting position, 40" standing.
-Eyes can only focus on 1 moving object at a time.

By adding the MANSTAND and MANSIT components to this drawing, we can make the following observations:

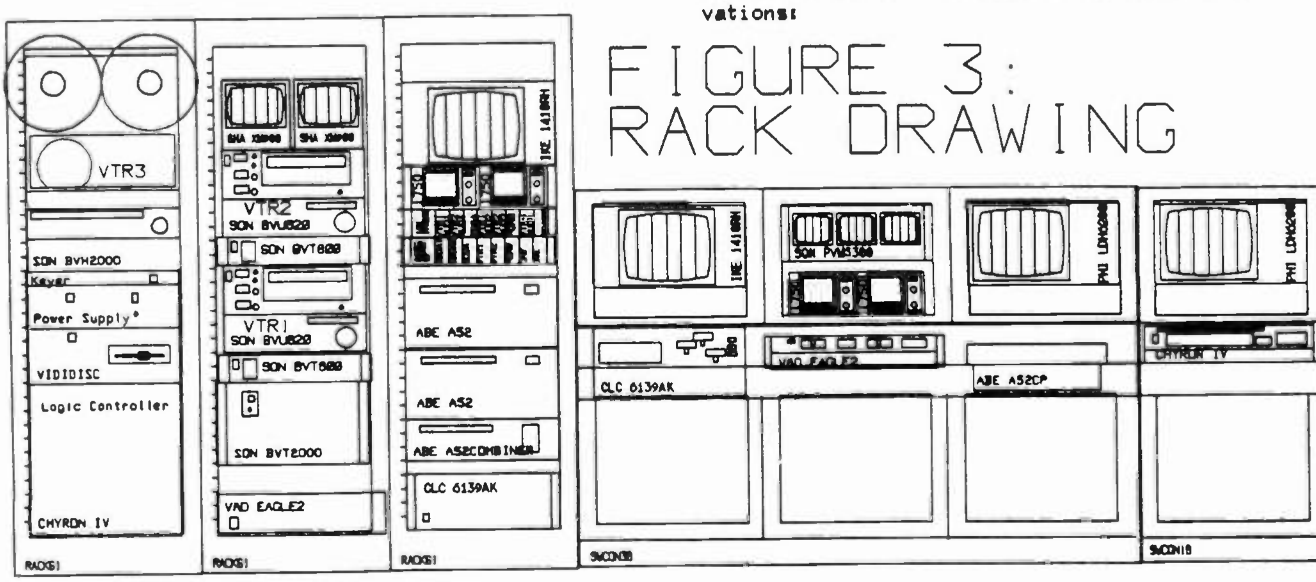


FIGURE 3: RACK DRAWING

-It will be difficult to change the tapes on VTR#3, so careful thought must be given to the frequency of tape changes. If tapes will be changed only every hour or so, this placement will be OK, but it would not be appropriate for spot tapes.
-The monitors in RACK#3 are too low. As a general rule, the center of any monitors that are to be viewed often should be on a line 15 - 20 degrees lower than your eyes.
-Good monitor placement in the switcher consoles.
-It will be impossible for one operator to operate the consoles as designed. Try an organ console.
-TBC#1 and #2 are well placed for periodic adjustments.
-Higher racks could be used for monitoring, especially if larger monitors are used.

ERGONOMIC SIMULATIONS

Because of the detail given in our drawings, the ABC Olympics engineers have adopted VidCAD for life-size ergonomic simulations. They plot our face and top views in real scale and place them on plywood frames. In this way operators can simulate the use of components to see if the layout is appropriate. From the engineer's vantage, CAD allows them to much more quickly plan and perform the simulations.

ARCHITECTURAL FLOORPLANS

CAD makes it pleasurable to do floorplans, because you can make corrections and move equipment around as you desire. Figure 4 was done with the parallel line drawing mode found in advanced drafting CAD programs. There are too many racks in the main room. We will move these into the adjoining room. You use floorplans for:

- -Set layout and design
- -Designing cable trays
- -Cable measurements
- -Electrical layout
- -Lighting placement

CALCULATING CABLE LENGTHS

There are 2 differing schools of thought regarding cable measurement: Fixed Length or Variable Length schools. Each approaches cable measurement differently, as shown below.

Fixed Length proponents believe it is not necessary to calculate every cable run. Under RS-170A specs, it is usually advisable to have equal length SEG supply cable runs, at least from the Null Point, usually considered the DA's. Consequently, it is critical to find the longest cable run--a job effectively delegated to CAD.

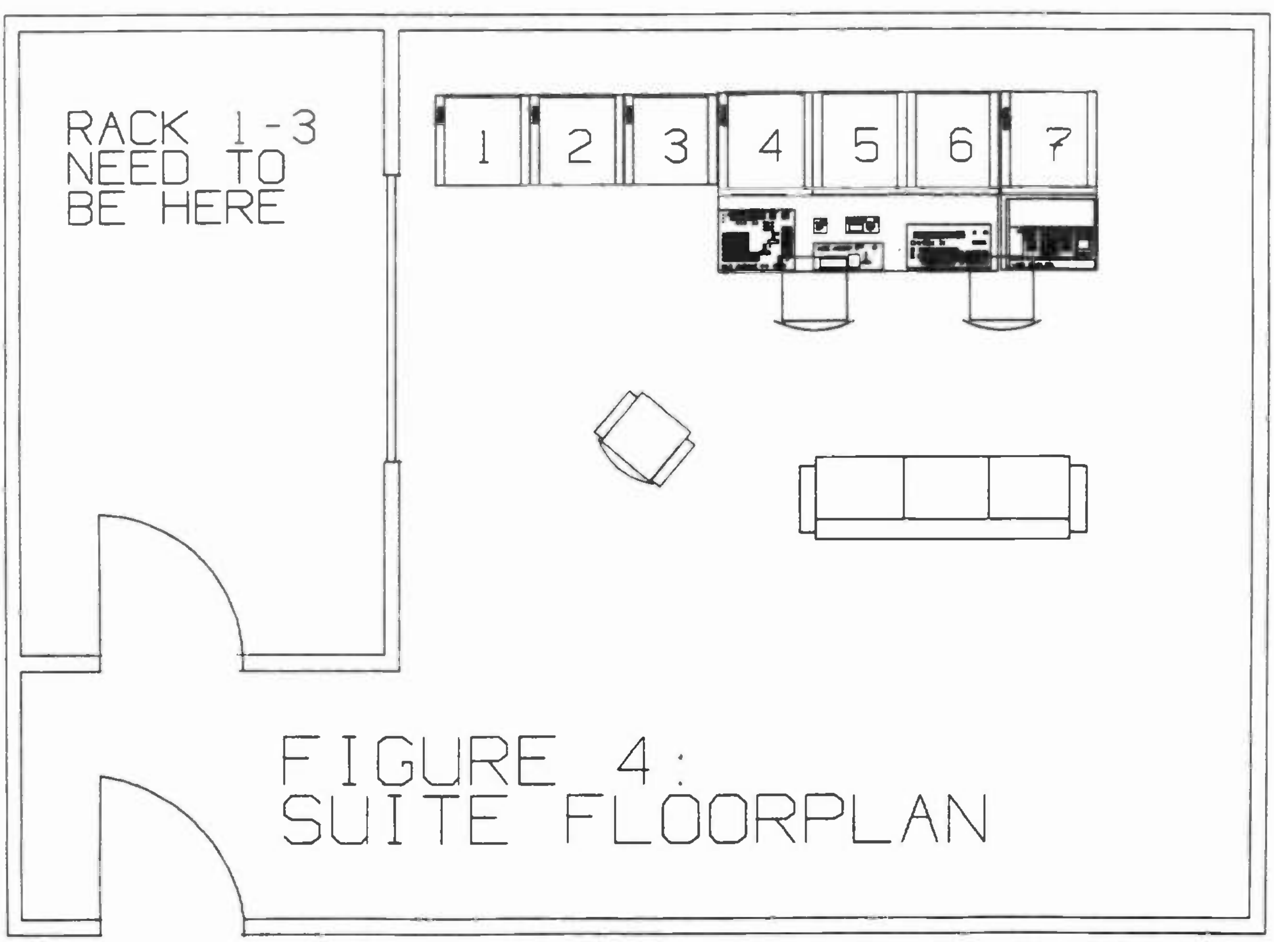


FIGURE 4:
SUITE FLOORPLAN

Variable Length proponents prefer to measure each cable run and bundle them because (1) contemporary equipment has adequate genlock capabilities, and (2) bundled cables are more aesthetically pleasing. Without CAD, measuring these runs before construction was a difficult and time-consuming manual task. Now it is much easier.

Whichever school you belong to, it is necessary to measure some cable lengths. It is also necessary to find the lengths of the remote control units' cables. Since there is a growing trend toward remotable equipment, and this cable is so expensive, difficult to cut, and even more difficult to stretch, using CAD to help you measure the correct lengths can pay for CAD!

Another way this excercise can help during installation is in estimation of delay, especially due to external Chromakeyers. By doing this accurately before installation, it can help you decide on the most efficient delay strategies (filters or cable), and enable you to more closely order the appropriate filters, cables and compensating DA's. This can save both time and money during installation. In fact, CAD systems can often save enough time and money in this one area to pay for themselves in the first job--especially in multi-core cables and delay lines.

Cable Measurement Tips

To start, make yourself a table like the one below (Figure 5). This same table can be used for a permanent CABLE MASTER TABLE.

a. For all components within the rack units, Measure Distance from component to component. You must have already decided how you will run cables between the racks. If you are running them to the floor, into a cable tray, and then up to the next component, you should start the measurements from the farthest point in the component down to the approximate level of the cable tray, then up to the farthest point on the other component. Enter this length in the table above under Rk-Rk'.

b. For all components connecting to the switcher console from the racks, using the above table, list the Rk-Flr' measurement from the farthest point on the component to the bottom of the cable tray. Then go to the floor plan drawing and MD from the farthest point in the rack to the farthest point in the switcher console. Remember to follow the path of the cable tray.

c. After totaling the Rk-Flr' and Rk-Rk' measurements, add in the "Rack-Factor," which is used to compensate for slight cable run bends during lacing, plus allows for pull-out of rack slides. You may need to use a larger factor, depending on your installation methodology. Components mounted on suitable chassis, such as SEG remotes and keyboards, will need larger factors. One exception to the 3' rule is for DA I/O's. Since they are usually in a group and are not put on rack slides, you may be able to use smaller cables.

SYSTEM FLOW DIAGRAMS

CAD significantly speeds up this most difficult task. It allows you to simply place the components in a logical fashion on the screen, then make the necessary cable connections. From this, it is easy to develop a complete Bill of Materials, including cables and furniture.

Perspective

It is very difficult to place the entire system on one drawing because it would be so small that even the largest plotters would render them totally unreadable. Yet, it is helpful to see the whole picture in a glance. Why not make 2 drawings: (1) Flow Overview--made from smaller (in DOS) front and top views; and (2) Detailed Flow--made from block views?

Flow Overview Drawing

This drawing is constructed from front and top views, each rescaled to be of similar size, making the drawing both readable and pleasing. I/O lines are only connected in general form to each component, without labels, to make it easier to read. Some tips:

a. Components Size. Each component should be a similar height on the drawing. This may be accomplished by changing the Component Scale. Or, it can be done with Component Rotate as was done on the DA's. Components may be derived from previous drawings.

b. Line Conventions. Inputs should be connected on the left and outputs should be connected on the right.

c. Line Colors and types. These are dependent upon your printer or plotter and video card. Here is one suggestion:

FIGURE 5: CABLE TABLE C

CABLE #	COMPONENT1	LOCATION	CONN	RK-FLR'	COMPONENT2	LOCATION	CONN	RK-FLR'	RK-RK'	FACT	TOTAL'	CABLE LENGTH
VTRDA3	VTR3-BVH2X	RACK1	BNC	4	VDA3-PDA37	RACK3	BNC	4	13	6	27	50
										6		

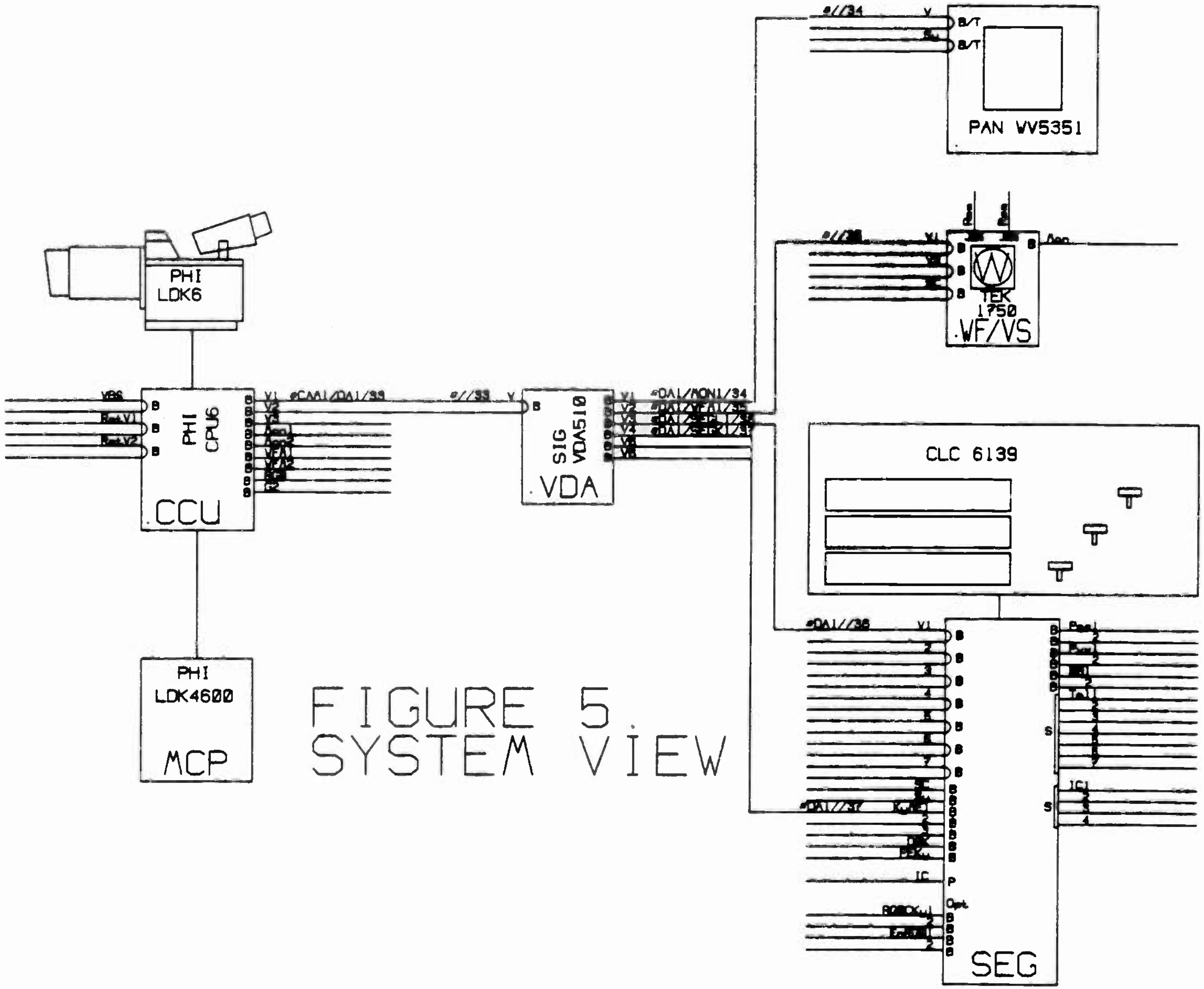


FIGURE 5
SYSTEM VIEW

Line Type		
Component Drawing	=	1
Video Line	=	2
RGB Line	=	3
Component Video	=	4
RF Line	=	5
Mic Audio	=	6
Line Audio	=	7

Color		
Component Color	=	1
Video Cable Color	=	2
Audio Cable Color	=	3
RCU Cable Color	=	4
RGB/Component Video	=	5
Video/Audio Combo	=	6

d. Component Groupings. To save space in the system overview, only draw system groups once. For instance, a normal studio may have 5 VTR's, each with TBC, DA, monitor and waveform monitor. Make one component containing all of these and place it once on layer 1-5, but on top of the first one. Your Bill of Materials is accurate, and you have not cluttered up the drawing.

e. Component Placement. Arrange components and modules in a meaningful way around the central point (e.g., SEG or Master Control).

Flow Detail Drawing

Detailed drawings are necessary to show the complete input and outputs, to ensure all components are properly connected, and to extract documentation. These drawings are also essential for proper system installation. Here is a sample of a detailed flow drawing (Figure 6). Note how easily cable lines join component output lines. This is because of the following conventions:

a. Component Placement
-Components are only one color, as color has been reserved for lines.
-Snap to Grid should be set for 1", as all components and lines are spaced with this. Consequently, it will be easier to space type and lines.
-It is helpful to Zoom Window to each component and mark the lines you are connecting with a Standard Point. Then you can connect lines to these points using Nearest Point while you are Zoomed.
-Consider placing each major subsystem on different layers, and each type of line on a corresponding layer. For example:
Major Video components are placed on Layers 10, 20, etc.
Minor Video components are placed on Layers 11, 21, etc. Video Lines are placed on Layers 12, 22, etc. Sync Lines are placed on Layers 13, 23, etc.
Major Audio components are placed on Layers 15, 25, etc.
Minor Audio components are placed on Layers 16, 26, etc.
Audio Lines are placed on Layers 17, 27, etc.
Mike Lines are placed on Layers 18, 19, etc.
If you use this, or some similar structure, then name each component using the layer name (e.g., VTR20, TBC21). This will allow you to quickly identify which layer is to be shown.

b. Cable Label Tips
-Text Size (TZ) = .75" or 1.0".
-Turn off Grid, or use 0.1".
-Text Rotate as needed.
-Text Place just above each I/O line for legibility.
-Begin each cable name with a "#" (necessary for automatic documentation. Also use a standard means of identifying your source, destination name (e.g., #VTR2/TBC2/L1 or #BVU870-1/BVT810-1/DOC).

c. Save Batch. Save each layer as a drawing, and save all layers as another drawing, plus save total drawing if you desire (but only if the layers do not overlap). REMEMBER TO ALSO SAVE THE COMPOSITE LAYERS USING SAVE BATCH (SB) so that you can use the *CADLIST* Bill of Materials and *CableDOC*.

CABLE DOCUMENTATION EXTRACTION

By storing the drawings in Batch mode, we have created a group of programs we call *CableDOC* to automatically:

-Extract documentation from the Batch Drawing
-Enter documentation into the *CableDOC* database

This is a sample of a *CableDOC* printout (Figure 7) for our system drawing. You can place as much of this information in the drawing as you desire, or you may only place source/destination/cable name information, and enter the rest outside the CAD program.

Here are some of the uses for *CableDOC*:

-Maintain cable name master log
-Identify and update source and destination
-Enter miscellaneous installation and maintenance remarks
-Generate run lists for installers and engineers
-Print cable labels without additional software
-Keep updated cable documentation
-Lock-out cable name duplication

CableDOC comes complete with several different output formats to automatically print labels on several common Brady computer cable markers.

CALCULATING AIR CONDITIONING, POWER AND WEIGHT FROM DRAWINGS

By combining the extensive SEARCH & COMPARE database of products listed by type, features and specification to *VidCAD*, we have been able to design a powerful tool we call *CAD Spec*. Essentially, *CAD Spec* extracts drawing information on all components in a rack and calculates the BTU's, power consumption, and weight automatically. It does not yet add this back into the drawing automatically, but it is an easy task to type these 3 numbers into the drawing documentation.

ATTACHMENT 1: PC HARDWARE CONSIDERATIONS

Here are some minimum hardware packages and average pricing. IBM may cost more while mail-order systems may cost 40% less.

Minimum Configuration--PCXT

PCXT (turbo recommended) with 640k memory, co-processor (essential for XT)	$1000
Hercules graphics monitor and card	$ 300
Dot matrix graphics printer	$ 300
Mouse	$ 200
TOTAL	$1800

Better Configuration--PCAT

PCAT 8 MHz, 0 wait states, co-processor, 640k memory	$3000
20 Meg hard disk	$ 800
EGA or PGA card and monitor	$1300
Graphics tablet	$ 600
AB plotter	$1000
Dot Matrix graphics printer	$ 300
TOTAL	$7000

Optimum Configuration--PC386

PC with 80386 16 MHz processor, 1 Meg ram, co-processor	$6000
40 Meg hard disk	$2000
Tape backup system	$2000
1024x1024 HiRes CAD monitor and card	$5000
Graphics tablet	$ 600
CD size plotter	$5000
TOTAL	$20600

ATTACHMENT 2: WHAT TO LOOK FOR IN CAD SOFTWARE

The following table has a few of the minimum features you should look for in a PC CAD system:

- -Zoom (to select a Window and zoom in or out to see detail)
- -Snap to Grid (to place lines and components on predetermined grid spacing)
- -Measure Distance (to precisely measure distance between 2 or more points)
- -Absolute Coordinates (to see the true X & Y coordinates on the screen)
- -Relative Coordinates (to see X & Y's relative to the first point)
- -Dimensioning (to precisely measure distance and automatically add witness lines and measurements to the drawing)
- -Windows (to mark a rectangle and zoom, erase, move or componentize contents)
- -Parallel lines (to draw 1 line and construct a parallel line x" away)
- -Edit (change drawing base line, colors, layer placement, components, line break, move, and erase)
- -Layer control (to select electronic overlays and draw on them. This is especially nice for complex drawings, because you can draw different detail on separate layers, showing only what you want at a time)
- -AUTOCAD DXF drawing interchange capability
- -IGES mainframe CAD drawing interchange capability
- -Support for most video boards and monitors--especially yours!
- -Support for most common plotters and dot matrix printers
- -Floating point based data (6 digit precision is helpful)
- -Numeric co-processor support (essential for speed, critical for an XT)

Quick Summary of Selected PC CAD Systems

AUTOCAD--40% of 1st time CAD buyers select AUTOCAD. It has hundreds of applications available, and it is well proven. However, it is not easy to use, especially if you do not use it regularly. We recommend AUTOCAD for full-time draftsmen and those already familiar with it, or for those who need a special application in addition to video and audio CAD.

VERSACAD is easier to use than AUTOCAD, but does not have the market penetration.

MICROCADAM is an excellent true 3-D CAD program, and interfaces directly with mainframe CADAM. However, it is expensive and does not interface with any other CAD system.

GENERIC CADD is an inexpensive CAD program with most of the AUTOCAD features at much lower cost. It is very easy to use, interfaces with most hardware, and has superior Batch capability. It is currently limited in drawing size, but none of our customers have made a drawing too big for it yet. It can convert files to AUTOCAD or IGES.

Other CAD Systems--There are hundreds of other CAD programs of varying cost and quality. Our approach is to support 3 drawing standards which can be translated to most other CAD systems: International Graphics Exchange Standard (IGES), DXF(AUTOCAD transfer standard), and Generic CADD. These 3 enable our application to be translated to probably 90% of existing CAD software.

INTEGRATING DIGITAL COMPONENT SYSTEMS INTO THE ANALOG AND HYBRID BROADCAST PLANT

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ABSTRACT

This paper will outline system considerations when interfacing 4:2:2 Component Digital based technologies into existing analog hybrid or all digital facilities. A short review of relevant protocols pertaining to interfacing hardware to conform with the given standards will be discussed. The next section will describe the technical considerations and solutions when implementing 4:2:2 based products into existing facilities. The remaining part of the paper will discuss future considerations when implementing component digital technology within an all digital complex. This may imply a divergence of operating methodology from today as well as peripheral support equipment. Also, new standards may affect the user-friendliness of the equipment as well as their communication both internally and to the outside world.

1. REVIEW OF RELEVANT PROTOCOLS

The recent approval and standardization of the 4:2:2 Component Digital Format for a DVTR is essential for progressive efforts towards the digital production studio. The format is unique in two respects: First, it allows for the design of the mechanisms and signal processing systems to be used worldwide for any digital television signals that conform to CCIR Recommendation 601. Second, the designer of a D-1 tape transport mechanism can use one of several different combinations of tape scanner diameters and data head arrangements. The format permits different design choices for different applications. However, implementation of the hardware must be in accordance with the guidelines set forth by the respective protocols to allow compatibility between systems. The following is a short review of the relevant protocols pertaining to the interfacing hardware conform to those standards.

a. CCIR Recommendation 601

This recommendation specifies the basic parameter values for the 4:2:2 digital standard. The format specifies that the Y, R-Y, and B-Y signal components are to be formed separately and encoded using the internationally agreed upon sampling rates based upon a 4:2:2 ratio. Hence, the Y channel is sampled at 13.5MHz and each of the color difference components (R-Y, B-Y) are sampled at 6.75MHz.

b. SMPTE RP-125

This practice describes a bit-parallel undirectional digital interface for component video signals, meeting the requirements of CCIR Recommendation 601. Encoding Parameters of Digital Television for Studios. The interface is applicable for 525/60 Systems M digital television equipment.

The video signal is transmitted in a parallel arrangement using eight conductor pairs. Each pair carries a multiplexed stream of bits of each of the Y/R-Y/B-Y signals. A ninth conductor pair carries a clock signal at 27MHz. The signals on the interface are transmitted using balanced conductor pairs for a distance of up to 50m without equalization and up to 300m with appropriate equalization.

c. EBU TECH 3246

This specification for a bit-parallel unidirectional, nine pair interface is functionally equivalent to the SMPTE RP-125 document. However, the specification is for systems operating in the 625/50 environment and conforming to CCIR Recommendation 601. The only addition is the proposal to allocate two lines explicitly for the transmission of auxiliary signals. The data signals are time-multiplexed and transferred as an NRZ code. The signals consist of video data, timing reference, ancillary and identification signals.

d. ANSI S4.40-1985 AES/EBU BUS

This document describes a serial digital interface for the transmission of digital audio signals between digital audio systems. The interface is designed for the transmission of one, two or four channels of digital audio over either a pair of wires, or an optical fiber, in a serial format. In addition to digital audio channels, the digital interface also permits the transmission of information related to the channels. These are user-definable data, information on the interface itself, error protection and additional digital audio channels.

e. Serial Digital Interface - 243Mb/s and 216Mb/s

Presently, there are two proposed serial transmission schemes to allow interfacing between systems conforming to CCIR Recommendation 601. These interfaces allow for the serial transmission of video data between systems through either coax or optical fiber. The proposed schemes are either a CCIR recommended encoding format based upon an 8B-9B Bit-mapped block encoding technique with a transmission rate of 243Mb/s or Sony's proposed coding scheme using NRZI at 216Mb/s. Both serial interfaces can be implemented using existing LSI technology which will make interfacing both economically and physically viable.

2. APPLICATIONS AND CONSIDERATIONS WHEN IMPLEMENTING 4:2:2 TECHNOLOGY

The Component Digital VTR will find many diverse applications in the marketplace. Since the 4:2:2 format was adopted, many manufacturers have begun to market equipment compatible to the CCIR Recommendation 601. Equipment interfaces conforming to the protocols stated above will allow 4:2:2 based signals to be routed in digital form to minimize signal degradation loss. This allows for the transparent recording, manipulation and reproduction of video and audio signals.

The benefits of utilizing component analog video technology in the production and post-production environment versus an all composite environment have been well covered in the past. In considering the implications of an all digital facility, we should first review a component analog island as a basis to start from.

Referring to Figure 1, it can be seen that he heart of the system is the analog component video switcher/mixer. The VTRs as well as all inputs/outputs are component analog using either RGB or Y/R-Y/B-Y components. In considering the problems that may arise when integrating various products together, certain cautions and solutions take form. Two major concerns that plague most edit environments are related to timing and levels. These two problems, when addressed, take on a possible solution as depicted in Figure 2.

In looking at Figure 2, timing relationships are first reviewed. The main TSG supplies reference signals to the complement of equipment. A source synchronizing generator which locks to the master TSG and drives the video source, allows easy timing adjustments for the source with respect to the system. Normally, the device with the longest path length is taken as the reference device and delays are then introduced in other devices through coax or lumped delay lines. In addition, other attributable delays are introduced from coax length, D.A.s, and the equipment itself. Automatic delay D.A.s or isophasing amplifiers can correct some of the timing problems. However, the best solution is to plan carefully, accounting for all timing problems in the initial design stage.

In a component island, level matching between the three video components is critical. Luckily, with the continuing interest in component systems, a variety of component based test equipment is emerging to allow us to monitor both levels and timing relationships between systems. Should a composite analog signal be introduced or outputted, a high quality decoder or encoder should be used. Also, provision should be made for a component color corrector.

The evolution of 4:2:2 based technology will bring about many benefits in the production and post production arenas. Let us now consider a "Digital Production suite" as depicted in Figure 3. In using a component based digital system, the recording of components permits multigeneration processing without any degradation of image quality. Again, the heart of the system is a digital switcher/mixer. The digital interface between sub-systems can be either the standardized parallel interface conforming to RP-125 or EBU Tech 3246 or the proposed serial interface. With the availability of digital serializers in the near future, interconnection between systems will become easily attainable. The monitoring of signals would continue to be made in the analog domain; in component form. Within the near future, due to VLSI design, it may be possible to implement digital parallel or serial inputs/outputs to existing high quality monitors. Notice also that the digital color corrector plays an important role in the edit area. A corrector may be required for a playback DVTR and preferably one should be assigned to the switcher so that independent assignment to any source is available. The color corrector should correct for black and white levels as well as gamma and operated in RGB. Suitable "digital D.A.s' and an assignment matrix would be required, as in the analog case. It is also conceivable that the architecture of the switcher and the use of the DVTR's will change in the digital environment, due to the ability of the system to make recursive effects/edits without picture impairment.

Again, acquisition of material from the composite domain should be kept to a minimum and, when necessary, a high quality decoder used as depicted in figure 4. If a direct analog camera source is brought into the area, a digital converter box (RGB to RP-125) would be necessary. However, given the advances in technology, it is conceivable that this may be built into the switcher or other devices.

Another thought is that the camera can be fed into the RGB inputs of the DVTR initially since the video DVTR'S bandwidth will not limit the picture quality. It should be noted that digital outputs from telecines will soon be available that will conform to the CCIR Rec. 601 standard. In addition, many graphics and animation systems already conform to the RP-125 parallel standard so that transfers to and from the source can be kept at the first generation.

It is apparent that much of the equipment in the high end post-production areas utilize component digital techniques to create a high quality end product with many special pictures processing features. The end goal is to bypass the analog interfaces for the inputs and outputs and only use digital interfaces between systems. The real benefits of multi-layered effects, clean chroma keys, and minimal signal degradation using equipment such as computer graphics, paint systems, digital slide stores, and effects processors can then be realized.

In expanding the above category, we now look at Figure 5 which depicts an expanded view surrounding one piece of equipment. The analog inputs/outputs can be either composite or component. In addition, many manufacturers now make available combine both analog and digital interfaces. As discussed, with the advent of VLSI design, it is feasible that serial interfaces may soon become available and readily implemented into existing hardware or offered as external options. Due to long signal paths, the digital inputs may require timing adjustments to take care of synchronization problems. Automatic phasing systems are now being considered which will phase lock all incoming signals to be processed to the inherent reference incorporated within the digital signal.

Another area of concern is the use of the four available digital audio channels in the 4:2:2 format. With the progression of MTS broadcasting and the proliferation of stereo based software being made available, production companies are paying more interest in the field of audio. Presently, it is envisioned that the audio sweetening area will continue to be a separate function, which ultimately has the responsibility of sweetening and conforming the audio tracks to the final end product. However, with the digital audio technology available today for production, it is equally important to consider the significance of this medium in how it carries a product. The following are some considerations in applying digital audio technology to the hybrid or all digital facility. In the production stages of the shoot, if digital audio is to be used, it should be noted that the 48kHz sampling rate conversion is needed during transfer if the data is to be transferred in the digital domain. In both cases, whether analog or digital equipment is used, the systems should be referenced to master sync. During predubs or playback, the systems should be locked via a audio/video synchronizer that is referenced to house sync. The worldwide acceptance of the AES/EBU serial data bus will play an integral art in the digital facility. The bus will allow for the transfer of digital audio data between systems without any signal degradation. The DVTR should have provisions for advance digital audio data to be available with adjustable delays. This is needed if the audio is to be sent to a digital audio mixer for sweetening or that the audio is to be distributed over long runs and channeled through extra processing.

With the integration of audio into both video and computer based industry products, the role of audio in production and post-production will become increasingly important.

3. TECHNICAL CONSIDERATIONS

a. Timing and Reference Signals

As with analog technology, various parameters must be monitored, tested, and adjusted to conform to given contraints. This holds true in digital technology as well, and with the implementation of a hybrid or all digital studio, these guidelines should be adhered to. When the digital interface is considered, each piece of equipment must be examined carefully from a reference and timing point of view. Although various discussions have taken place regarding reference signals for the evolving marketplace, we must at present utilize existing reference signals. These signals take the form of mixed sync or black burst. New to the reference arena are digital signals with inherent timing references such as described by the RP-125 parallel digital interface.

Within the equipment, timing adjustments for critical processing blocks that relate to the outside world are important. Questions as to where and how the timing should take place and relationship to the reference or input signal must be addressed. The adjustments will either adjust the analog and digital parameters indepen dently or together. If the adjustments are global in nature, care must be taken in planning both the analog and digital paths independently. This is because the timing for the digital interconnects between sources may not be the same for their analog counterpart.

Audio timing relationships plays an equally important role in the facility. With the introduction of four digital audio channels and the availability of digital audio support equipment, timing and phase relationships need to be monitored closely. Care in the distribution and handling of signals as well as conformance to specific sampling rates will become a concern. Pitch changes relates to variances in the sampling rate and the need for sampling rate and pitch converters will also be needed. Finally, the audio chain must relate to the reference sync. This implies that throughout the production, care must be taken to allow for synchronization of all processing systems.

b. Level Matching

Level matching is a concern for component analog and digital systems. Levels should be matched and corrected before digitizing. If it becomes necessary during the course of the production to color correct a source, then there are two alternatives. One is to convert back to analog, do the correction and convert back to digital. The other will be addressed by the digital color corrector, thus avoiding generaton loss through the A/D and D/A chain.

c. Monitoring/Test/Adjustment

In general, most of the monitoring and testing will be in the analog domain. As stated earlier, many test equipment manufacturers see the viability of a component test system. The system can address both the 1/2" market as well as the upcoming digital component market. Some test generation equipment already have compatible digital parallel outputs that conform to the 4:2:2 standard. In addition, digital test devices will soon become available for monitoring and testing the DVTR. This may encompass an error rate checker and monitor to measure a DVTR's performance as well as other devices to check processing blocks. Other devices will check the performance outputs of digital serial feeds as well as monitoring timing, phase, and level relationships of signals. However, with expanding knowledge in digital technology, adjustments will be minimized due to the inherent stability and advantages of digital processing techniques and VLSI integration.

4. FUTURE POSSIBILITIES IN DIGITAL COMPONENT EDITING

The digital VTR, due to its ability to store and reproduce transparent images, will become a cornerstone in the foundation of a new revelation in broadcasting and production. With the flexibility of digital signal processing implementation, editing islands of the future may instill more creative freedom. This may imply a divergence of operating methodology from today as well as peripheral support equipment. Also, new standards may affect the user-friendliness of equipment as well as their communication both internally and to the outside world.

The improvement in quality of the end product due to digital implementation is impressive. Digital production techniques also saves steps in the production chain as compared to an analog equivalent. The following are some examples of what may change when using digital production techniques.

- Various operations can be performed without loss of quality: multilayering, processing, and recording.

- Stage or on-location shooting is simplified through the use of one camera since it will be easy to modify the image to correct for lighting and colorimetry.

- Downstream chroma keying on recorded material will be commonplace. This may take the form of post production chroma keying from a DVTR source.

- Effects generation with no loss of quality during transfers.

- Automatic phasing of source and processing signals.

- 2-D and 3-D image synthesis systems with direct digital I/Os.

- Composition of complex images with various sources in which frame and perspective transitions can be controlled with minimal effort.

- Possibilities for greater artistic freedom in all stages of production.

The future digital studio may change both the design concept of the studio as well as its intended use. Since each DVTR can transparently store and reproduce images in the digital domain, their applications will become more diverse. Digital switchers may include digital chroma keying, color correction and picture manipulation systems inherent in the system. New forms of graphics and animation outboard equipment will enhance the post production environment. In addition, editing systems will become more user friendly with many memory configurations.

Presently, graphics and animation production systems already exist that will interface to the 4:2:2 standard as shown in Figure 6. These systems augment the post production environment with the ability to generate graphics and animation as well as controlling the VTR's. However, in the future, it is feasible that these systems will become more intergrated into the post environment or become a separate fully self-contained production turnkey system.

5. CONCLUSIONS

The agreement of a worldwide standard for the component digital format is a great achievement for both broadcast endusers and manufacturers. The standardization of the format allows manufacturers to competitively design and manufacture a new breed of equipment technology. However, interfacing considerations must always be taken into account. This paper has outlined some of the technical considerations in interfacing 4:2:2 Component Technology into existing analog hybrid or all digital facilities. In addition, comments were given to the perceived problems in implementing this new technology into future edit suites. Finally, examples were given to show the advantages in doing a production in the digital domain.

The 4:2:2 standard launches a new era in broadcasting's history. The marriage of digital video and audio technologies brings about a new revelation in the manipulation, reproduction and transmission of audio and video signals. As a result, the broadcasters, production houses, manufacturers, and endusers can now consider the implication and possibilities of the all digital production facility.

FIGURE 1 ANALOG COMPONENT EDITING

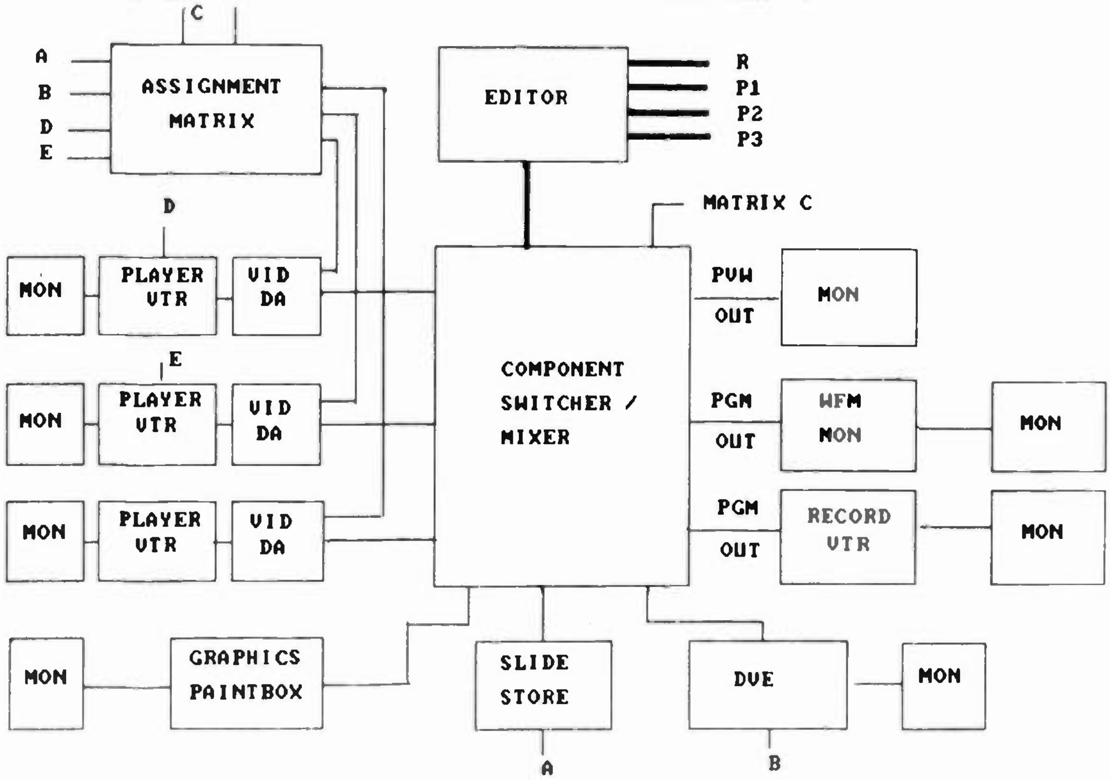


FIGURE 2 TIMING FOR ANALOG COMPONENT EDITING

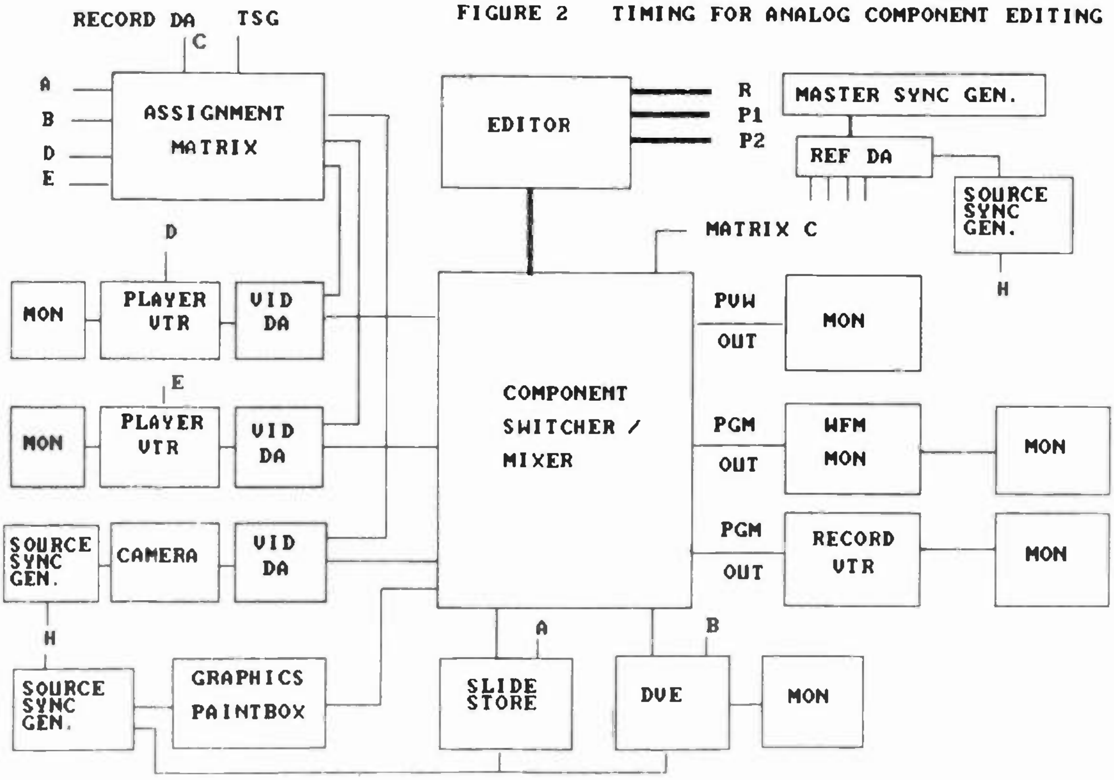


FIGURE 3 COMPONENT DIGITAL EDITING SYSTEM

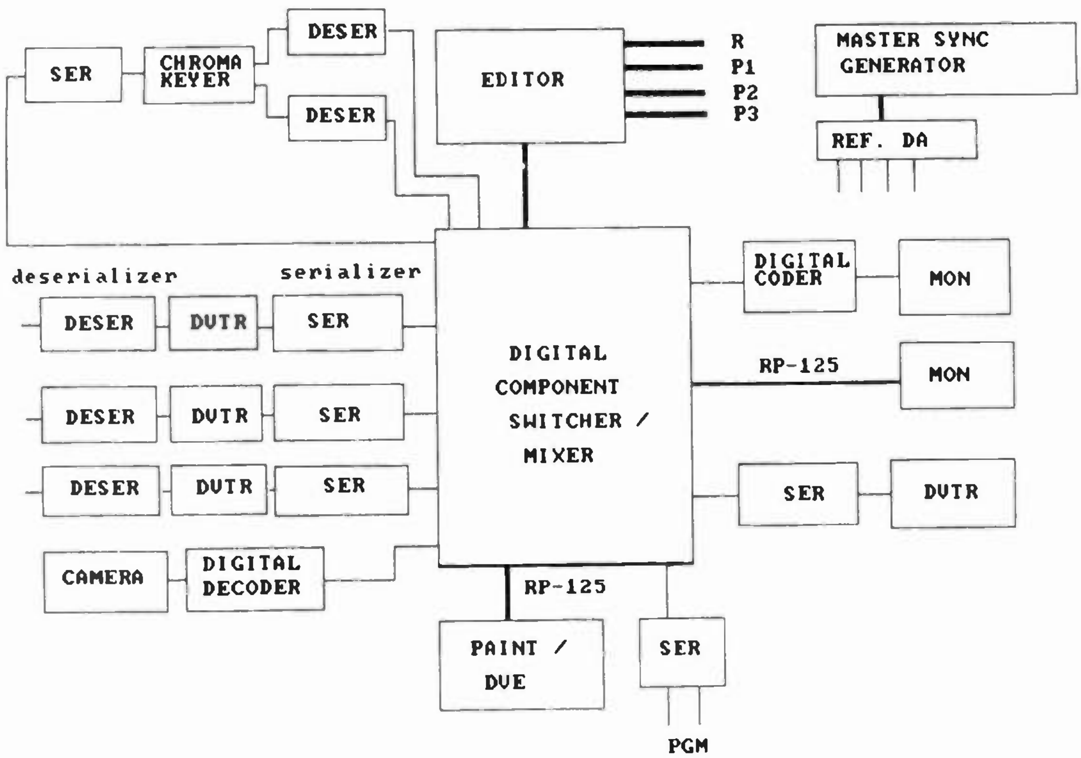


FIGURE 4

DVTR IN VIDEO PRODUCTION LAYERING / MASTERING

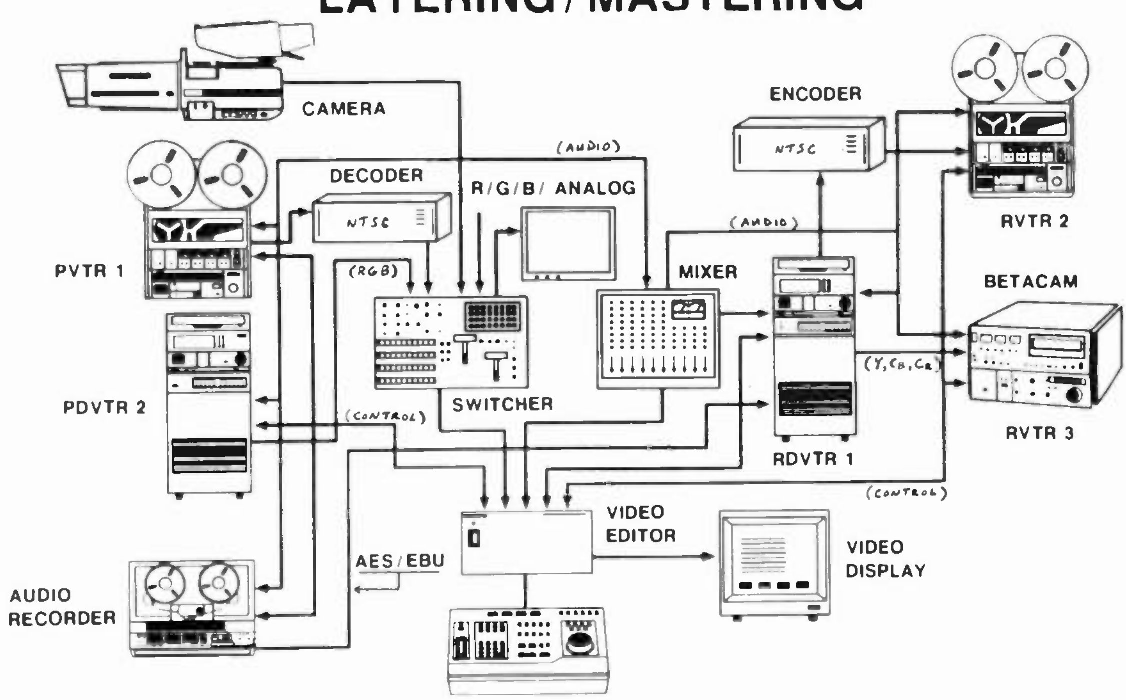


FIGURE 5 TYPICAL DIGITAL PROCESSING DEVICE OUTLINE

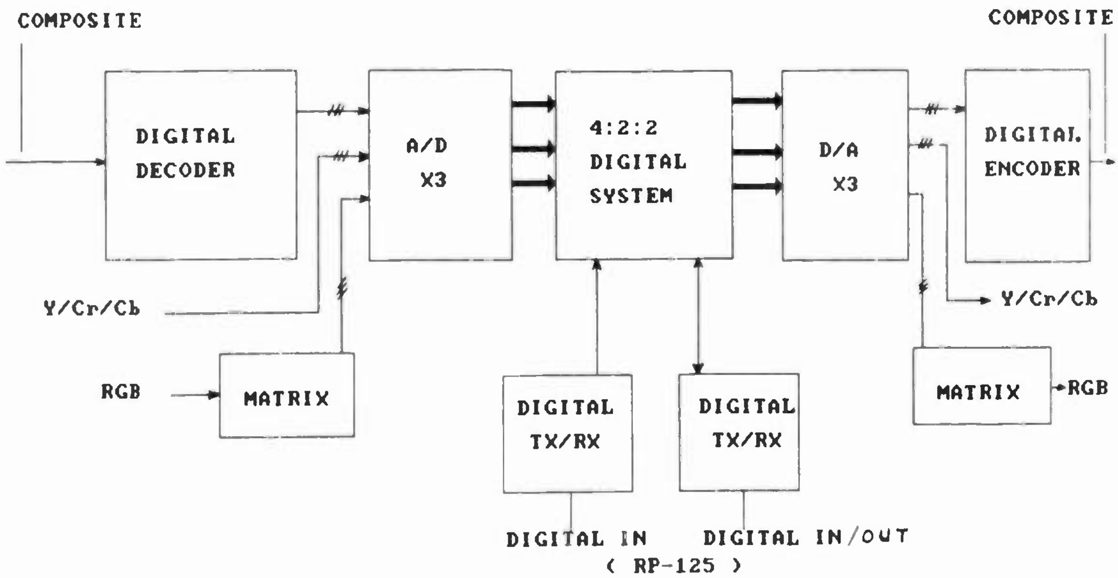
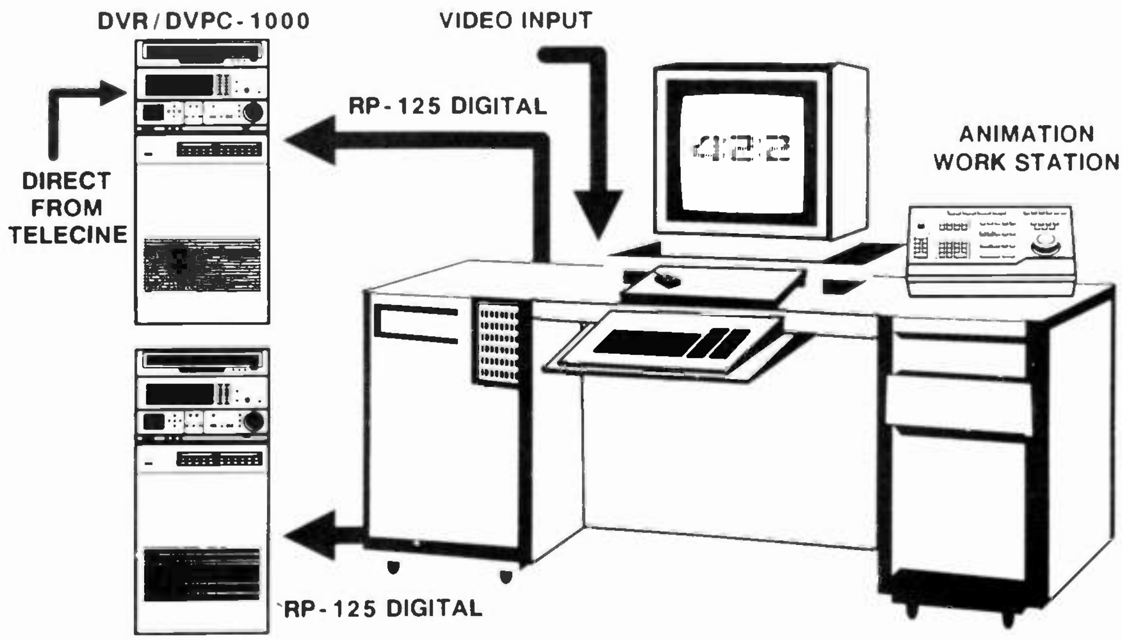


FIGURE 6

DVTR IN ANIMATION & PRODUCTION EFFECTS EDITING / ANIMATION STORAGE



FIBER OPTIC DEVELOPMENTS PROVIDE HIGH QUALITY VIDEO TRANSMISSION

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ABSTRACT

Far from being an exotic medium, fiber optic video transmission is a cost-effective, expedient means of signal transfer in any environment. Yet, due to its relative youth, its acceptance has been limited to those applications managed by persons having an eye for the future. With emerging signal source technologies comes greater acceptance of modern communications methods, and also a need for enhanced capacity media. This acceptance clears the path for fiber optic technology, providing for the needs of today and requirements of the future.

HISTORY

The rank and file of broadcast professionals have yet to embrace fiber optic technology as a medium to be exploited in the studio-transmitter plant. Apparently, technical misperceptions on the part of engineering staffs as well as apprehensiveness on management's part cause this condition. Without doubt, the heyday of fiber saw high prices, a by-product of low availability. The Law of Supply and Demand applies to high technology and this case was no exception. Telecommunication concerns, seeing the long distance benefits of fiber optic systems, released the reins on manufacturers. Large quantities of fiber needed to upgrade the Nation's telecommunications network, bolstered by the deregulation in the long distance industry and increasing demand, caused a bevy of entries into the field of fiber suppliers. By so breaching the common Catch-22 scenario of technical product introduction/acceptance cycles - and providing a revenue base - the industry flourished. Quality control in the fiber manufacturing process became part of an industry routine to ensure competitiveness. Packaging refinements, influenced by hundreds, then thousands of installed route miles, made new fiber installations easier and cheaper than their copper counterparts. Even more important was the lowering of the distance requirement for those installations to become cost effective. When the limitations of low durability were solved, fiber became almost a necessity in military installations due to its ruggedness alone. These applications enhanced progress in the development of real solutions to connectorization problems admittedly prevalent in the early years of fiber. Today's connectors can be field installed with the ease of common coaxial connectors, and at similar cost. Long gone is the need for specially trained technicians with a costly array of equipment, even in critical installations. Similarly, splicing, once considered a black art, is a task that even this multi-thumbed author can handle - and afford.

FIBER TYPES

There are three common types of fiber optic transmission cables in production. The most familiar to the lay person is the step index, multimode type. Often of plastic, it is the kind found in fiber optic lamps of the decorative nature and among those used in medical applications. These fibers are inexpensive, easy to package and connectorize, and have the ability to transmit light in the visible portion of the optical spectrum. On the negative side, a low melting point and technical limitations of transmission capacities (to be discussed later) limit the communications potential of this fiber family. The name stems from the manner in which light propagates through the medium; the indices, or paths, which the light passes through are differentiatiable into measurable steps. Thus, "step-index". Another type of fiber is graded index multimode. The most common kind of short range (under ten kilometers) communications fiber, graded index has technical advantages over step-index which allows use in critical applications yet maintains the cost advantages of inexpensive connectors and optical devices used for multimode fiber. It is still relatively large in size, allowing easy connectorization and splicing. Various subgroups of this category are organized by fiber size or loss characteristics, and further by packaging. The fiber itself is a small percentage of the diameter of the cable, with the bulk being protective coatings and strength materials for durabilitv and ease of installation.

The third type of fiber is single mode, sometimes called monomode. The name refers to the fact that in these fibers, light propagates through in only one mode, or path, instead of reflecting and refracting in different paths. Famous from advertisements of the long distance companies, this is the fiber that most lay people remember when they hear of fiber optic communications. Single mode offers the huge bandwidths and low losses often quoted by industry analysts and stockbrokers when touting the growth potential of fiber technology firms. Yet, for all its benefits, the bulk of short to medium range installations and many of those longer than fifteen kilometers are multimode fibers. Why is this?

There are several reasons. Single mode fiber is quite thin, about eight microns in diameter for the active part of the fiber. (The average human hair is 60-80 microns across.) Consider how critical it must be for a pair of these fibers to be aligned end to end to effect a splice, or connection to a receiving or transmitting device. Further complicating the matter is the need for a light source capable of delivering a useful amount of light (200 microwatts) into such a small diameter fiber. These factors nearly dictate the use of lasers for transmitters and special, high quality connectors in the fiber installation itself. While the fantastically high communications capability of single mode fiber in many cases justifies this effort, it is not always necessary. However, the number of so-called local loop applications of fiber using single mode is on a rapid increase due to the drop in cost and complexity of transmission equipment. All predictions indicate that single mode systems will constitute the bulk of fiber installations by the end of the decade.

NECESSITY FOR FIBER

The elimination of egress, or undesired leakage of signals, from metallic cables is a major concern. Witness the many stiff fines imposed by the FCC on CATV system operators whose lines leak signals, jamming two-way radios, aircraft, and the like. Armed Forces organizations use fiber optic lines to avoid detection by enemy intelligence. The fiber links can self-monitor and shut down if a tap or break is attempted, foiling a spy operation or other unauthorized monitoring. A fiber optic supertrunk, or multichannel television transmission system, provides a non-interruptable, interference proof alternative to microwave or copper links for the CATV system owner. The Army and Navy use fiber to circumvent egress leaks not only for security but to stop mutual interference between their many communications sytems.

In similar fashion, a system which leaks signals out is likely to let signals in. This is called ingress, and at the very least, can corrupt a data signal or television picture; at worst, it can mean collapse of a communications system and the resultant costs and damages. Metallic cables, by their nature, act as antennas for electromagnetic radiation. Fiber optics being nonconductive, cannot pick up any outside signals and are immune to these undesirable effects. Again important to military applications is the resistance to jamming provided by fiber optics. This is a natural side benefit of this inherent immunity to external interference, intentional or inadvertant. Fiber optic cables, compared to similar sized copper cable, are significantly lighter in weight, usually by a factor of ten or more. A several kilometer spool of four fiber cable suitable for conduit pulling can be easily carried by one man. Even if such an unrepeatered run of coax cable were possible, consider how much effort would be expended carrying it around and pulling it into place. Labor costs savings are part of the bargain one gets when using fiber. Since each of those four fibers has the potential communications capacity of many coax cables, with no need to boost the signals en route, the duct space crunch in many facilities is alleviated with fiber. Gone too are the expensive power supplies coaxial runs use to operate the in-line amplifiers so necessary for signal restoration.

Figure One shows a typical comparison between hard wire communications media and fiber optic bandwidth. The graph clearly indicates another of the benefits of fiber.

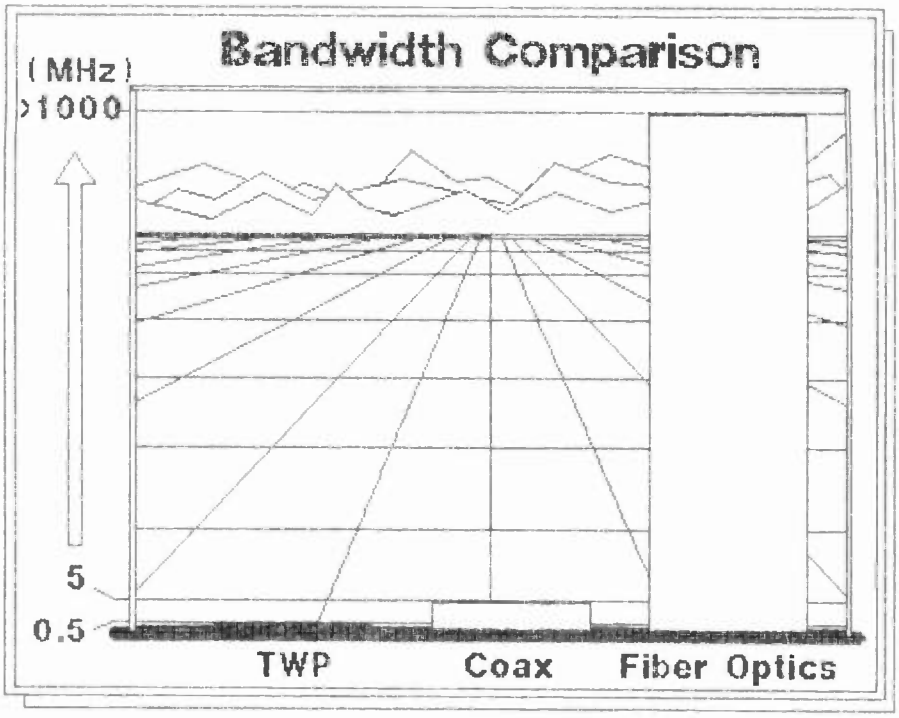

Figure One: Bandwidth Comparison between common media

TRANSMISSION TECHNIQUES

An optical source can be modulated at great speed. Some sources, such as the injection laser diode (ILD), can be modulated at greater bandwidths than the light emitting diodes (LED), which are more common and less costly. All types of optical sources are intensity modulated; that is, the power of the light output at any given instant is changing with the information being sent. In practice, the optical source has a quiescent state during which the light is on but not being modulated. The light is never blinking, so to speak, but varying in brilliance over a predetermined linear range. This range is different with each model of LED device and also among individual parts, and is a cause for distortion in low grade equipment. Even broadcast-grade transmission equipment can suffer from the effects of drift of this key parameter. An effective solution is the frequency modulation technique, where a carrier modulates the light source at a fixed fifty percent duty cycle and this carrier is frequency modulated with the desired waveform. When demodulated, the FM circuit will deliver a faithful reproduction of the original signal, provided that the equipment is of quality design. This method makes the vagaries of the optical link transparent to the communications taking place. If multiple signals, such as a composite of several television pictures are desired instead of one baseband signal, it is quite possible to make the aforementioned carrier quite high in frequency and create an FM-on-FM matrix not unlike domestic satellite transponders. Figure two shows a simple block diagram of the IM and FM schemes, with a view of the rarer digital method, explained later.

OPTICAL PROPAGATION

The two classifications of fiber optic cables, multimode and single mode, get their names from the manner in which the light information passes through, or propagates, in the medium. As previously mentioned, there are benefits to using single mode over multimode that in some cases outweigh the higher costs involved in that thechnology. If the disadvantages of multimode are not significant in a given application, then the dollar savings can be appreciated.

The reason that multimode fiber has less available bandwidth than single mode fiber is pulse spreading. Two contributing factors are the optical source and the fiber itself. The optical source for a multimode system is usually a LED. This device emits an optical signal that is spectrally wide in relative terms. A

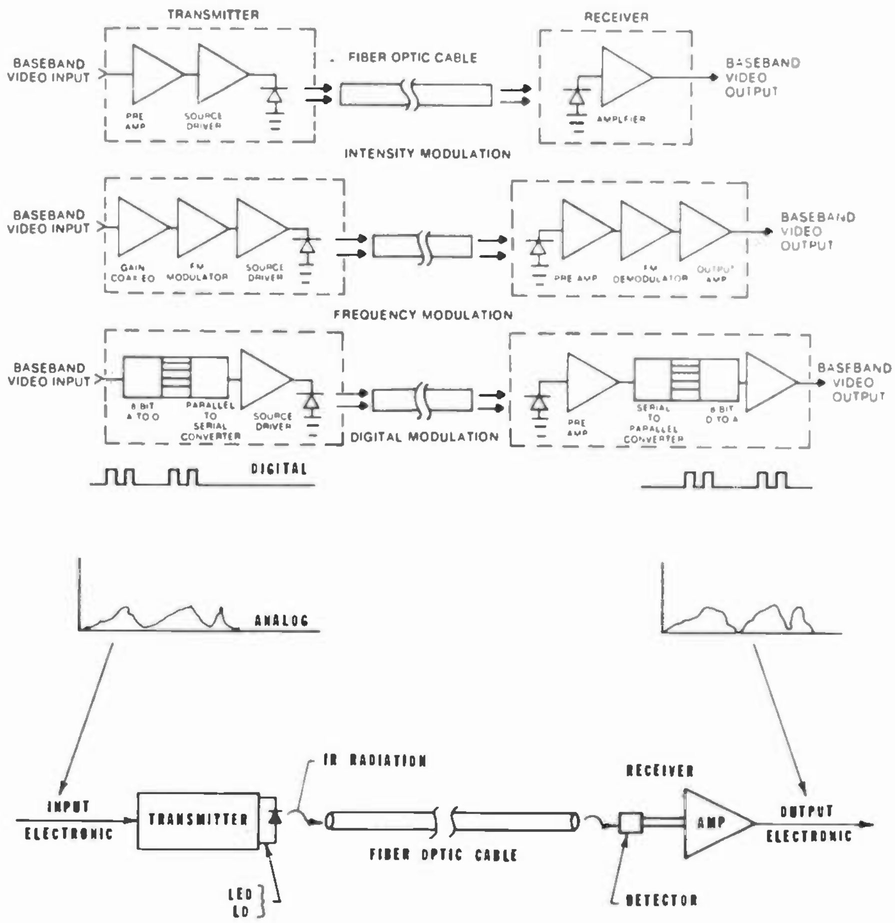

FIGURE TWO: Simplified System Block Diagrams, illustrating transmission methods

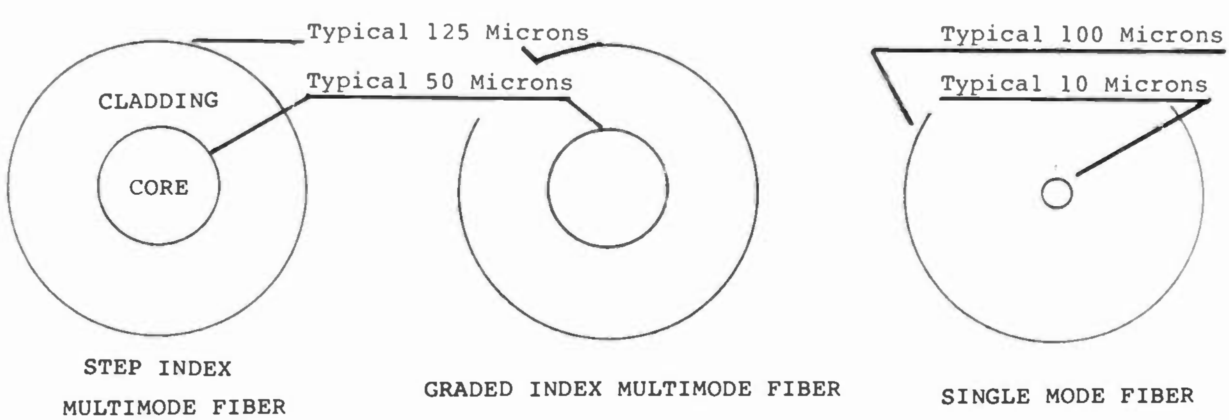

FIGURE THREE: Three Types of Optical Fiber typical example is a device with the bulk of its output power at 830 nanometers with a six decibel curve of plus or minus ten nanometers. The fiber itself has optical dispersion which limits bandwidth; indeed, the fiber is rated in megahertz per kilometer, with premium grade fibers commanding higher prices. When the phenomena of optical dispersion, similar to the effects of inductance, capacitance and resistance in coax, combine with broad spectral output of the LED, net available bandwidths can go as low as fifty megahertz per kilometer in some cases. Obviously, neither the distance or the bandwidth limitations of this example would hinder most broadcast applications. To demonstrate the advantage of single mode fiber, let's use the same criteria as we did in the LED example. The ILD which we shall use for single mode transmission has a spectral width of, say, four nanometers. And the single mode fiber connected to it has a rating of ten gigahertz kilometer. Therefore, the net available bandwidth in one kilometer would be nearly ten gigahertz, roughly the same spectrum as would be used by 1,666 television channels or 66,666 FM stereo stations.

An analogy that is seldom used is one comparing fiber technology with microwave technology. The comparison is usually ignored because a fiber cable is a physical medium and can replace coaxial cables largely because it looks like a coax in many cases. The microwave path, being invisible, does not lend itself to comparisons with fixed media yet this analogy is the best way to undestand the capacities, limitations and advantages of fiber technology. Equally valid is examining the dissimilarities of coax and fiber. While physically similar, the architecture of distribution dictates that one use a different mindset when configuring a fiber solution to a communications need. This point can not be more highly stressed and is discussed later in detail.

NEW DEVELOPMENTS

Available technology for fiber optic transmission systems is at once the limiting factor in its application and the reason for the commercial success of the medium. The most exciting technologies have been in existence for some time but the new developments in component manufacturing technology make them available at costs that will be acceptable to the end user. That spurs the acceptance of fiber and thus new developments in applications.

Injection laser diodes are available from numerous sources and at various performance levels. This is a relatively new condition as just two years ago this product was scarce and expensive. Now it seems everyone sells them and the price to performance ration is higher than ever. ILS's operating at the so-called third window wavelength of 1550 nanometers are now practical for use in super long distance, unrepeatered analog video links. Artel has achieved a record distance of 81 kilometers while maintaining EIA-250B short haul standards, using this technology. Manufacturers and customers alike can appreciate the tenfold increase in projected lifetimes of these components, especially when coupled with average power output increases of twenty percent in the past two years. Only safety restrictions prevent the use of even more powerful lasers in communications gear.

Another technology that has recently become cost effective is wavelength division multiplexing (WDM). Imagine, if you will, two or more different color lasers transmitting their signals down one fiber simultaneously. With the proper multiplex and demultiplex equipment, one could recover each different color light beam and send it to the appropriate receiver. It is even possible to send light in both directions at the same time. The benefit of this technology is the greater utilization of communications capacity in a fiber: if a fiber cost $500 to install and there are five communications services running on it, the cost of each is only $100 and all of a sudden, coax becomes a very expensive alternative. Beside being cost effective, WDM is easy to configure and use because of the small size and low loss parameters these devices have today.

While there are limitations when using a fiber optic technology, the major ones can be alleviated using optical devices that emulate their coaxial counterparts. This allows the configuration of fiber systems that mirror metallic cables. A caveat is that once you start adopting conventional architectures, the inherent benefits of fiber optic transmission are compromised and this compromise may jeopardize the very reasons for selecting fiber in the first place.

A passive tap, not unlike a coaxial tap, is possible to implement in fiber optics. It has a forward loss and a tap loss, each being a function of the other. In other words, to have a low forward loss, the tap would also have a high tap loss, not able to couple out as much light as if a higher through loss were permitted. This device would make possible an expansion of an existing point-to-point link or create a point-to-multipoint distribution system. With the proper electronics, a multiservice distribution network could be configured with fiber cable only. In the same way, a multipoint light coupler, called a star coupler, permits the use of the hub and spoke network architecture. These are particularly easy to configure.

The downsides of these "coaxial manifestations" are several. First, a distribution system using a passive tap on bus layout requires a high power optical source and installation of a different tap at each tap point to ensure an optical power balance at each receiver. It would be difficult to make an addition to the network once it is established, as each tap after the addition or deletion would need to be changed. Further complications include the cost of the high power source, the somewhat complicated installation and the potential for network failure. For instance, all communications stop if the common hub point is disabled for any reason, in the star network. It is also difficult to maintain security in these open architectures, it not being practical to monitor optical power levels in a system where those vary in normal operation, due to the high number of variable components. High losses prevent use in many applications where distances exceed the maximum permittable in the fiber systems. The bandwidth advantages of fiber are also compromised in certain multimode installations when that type of fiber is dictated.

Given these advantages and disadvantages over standard point to point applications of fiber, it is well to consider both approaches. When planning a system, the fixed-media microwave analogy serves well to illustrate the manner in which to configure the optical communications plant. Simply think of how one would effect a solution to the communications task at hand, using microwave. Assume, of course, no line of sight limitation! Then, replace all of the microwave transmitters and receivers with fiber optic functional equivalents. These re available from several reputable manufacturers. Given this method, any broadcast engineer can easily do his or her own fiber optic equipment application engineering with only minor assistance from the manufacturer.

DE-FACTO STANDARDS

There are some choices to be made among the individual types of fiber available today. Already discussed were the relative merits of multi- and single mode fibers. Among the multimode fibers there are several sizes produced. Each has characteristics, cost being paramount, that make them more or less attractive to the end user. Were it not for some default of choice by equipment manufacturers, there would be a state of incompatibility in installations. Fortunately, this is the case, and at least with communications gear that the broadcster would use, there is general agreement that the 50/125 or 62.5/125 size fibers are an accepted standard for multimode. The two numbers represent the fiber core diameter and cladding diameter, respectively: e.g., 50 microns core with 125 microns outside diameter. The radius would be 25 microns of core and 37.5 microns of cladding, in this example. Refer to figure 3.

This uniformity allows using standard connectors just as the uniformity in coaxial coables allows use of those fittings. Connectors for nearly any type of fiber can be supplied by many vendors due to this universality, and this situation encourages competition, improving quality and price. Optical semiconductor suppliers also vie for the business of transmission equipment manufacturers, who demand higher performance components to stay ahead of their rivals. All these factors, along with good performance and general agreement on interfacing, make the fiber equipment arena a buyer's market. Wise planners are taking advantage of this situation.

Digital video, and audio signals, are sent by sampling the analog waveforms at frequent, predetermined rates. This is the so-called Nyquist rate law, stating that the sample frequency must be at least as high as twice the highest frequency component in the sampled waveform. Such sampling results in a digital signal, consisting of a series of fixed value bits of data. Done properly, no degradation of the original signal occurs, and the digital nature of the resultant information allows for many retransmissions of the signal with no loss of integrity. The disadvantage of digital transmission is the high bandwidth required. To counter this, compression techniques allow digital signals to travel in narrower bandpass media, but at a sacrifice of fidelity. With the availability of bandwidth in fiber, the compression technology is unnecessary and full fidelity transmission is possible with all the advantages of the noise free digital routing method intact.

NEW APPLICATIONS

Entrepreneurs in every field lament that everything has been done before, that there are no new applications left. Well, almost every field. I have not heard people in fiber optics say "there's nothing left", and for good reason. Indeed, more new applications exist for fiber communications than the resources to develop products to address them.

Applications come both from within fiber optic equipment manufacturers and the customer. Of most consequence are those which customers suggest because they indicate a need, and in the case of the broadcaster, the need is usually urgent. If the number of potential applications warrant a rpoduct development, then rest assured that it will be created. A perfect example of this phenomena is the soon to be introduced SMPTE digital video standard. Once adopted, communications links will need to be made available that can handle this signal. Fiber optics provide the most sensible way to move the information around because of the bandwidth capacities of the medium. The high definition television (HDTV) signals of the future are similarly best handled by an optical transmission system, and in fact, one has been successfully tested recently. Long only the domain of the computer graphics industry, discrete red-green-blue (RGB) signals with their high bandwidth requirements are best sent long distances with fiber. These systems are critical over a fifty foot coaxial run but can be carried miles over fiber with no detrimental effects. New equipment is being offered that ties a high bandwidth matrix switch with optical technology to provide a routing and distribution system with fiber optic inputs and outputs.

With emerging technology acceptance and demand comes the future of fiber. Robotic vision systems are planned that use fiber instead of a camera to get images to a processing unit. This makes it possible to build a smart robot, one that reprograms itself to overcome unanticipated conditions. This surpasses the mouse in a maze artificial intelligence because it has application in the hazardous world of nuclear waste handling. In a similar manner, the military can use remote, intelligent drones to thwart the enemy rather than the more costly human alternative. Airline pilots to student drivers to space station dwellers will all benefit from stereoscopic, computer simulated vision in vehicle simulators of the future. These will make our transportation network even more reliable and safe. At home and relaxing, fiber optic developments revolutionize home entertainment, as optically transmitted data flows in the Integrated Services Digital Network. ISDN is even today being implemented, connecting McDonald's restaurant franchises together experimentally. Television, stereo recordings, computer and security data will someday flow to everyone via a glass fiber connected to the information network of the future.

And that future beckons...

Suggested reading for more technical discussions on fiber technology and the possible applications:

Kaiser, Gerd. Optical Fiber Communications. First edition. ISBN 0-07-033467-6; McGraw-Hill, New York, 1983.

Jordan, Edward C. Reference Data for Engineers: Radio, Electronics, Computer, and Communications. Seventh edition. ISBN 0-672-21563-2; Howard W. Sams & Co., Inc., Indianapolis, 1985.

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Klinger, Susan J., editor, et.al. "GuideLines". Telecommunications Products Division, Corning Glass Works, Corning, New York. 1983-87.

UHF MULTICHANNEL TELEVISION ANTENNA SYSTEMS

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ABSTRACT

Regulatory pressure is increasing to limit available site locations. The FAA restricts the locations for tall towers. With the EPA guidelines, local governments are increasingly more difficult to satisfy about the safety of new transmitting systems. The most economical solution is the use of a single, multichannel antenna and a single transmission line.

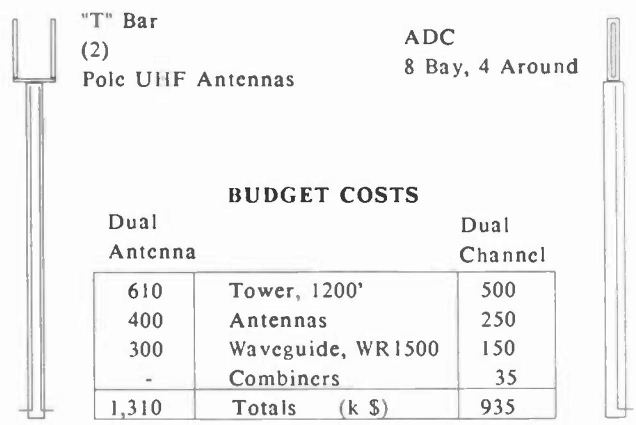
This paper discusses the advantages of using a single antenna and transmission line for high power, UHF transmission systems. The types of antennas suitable for broadband, multichannel application are presented. Antenna performance capabilities in terms of patterns, VSWR, and power handling are reviewed. The bandwidth limitations and design requirements of waveguide transmission line, components and transmitter combiners are covered. In addition, the system design and performance of a channels 50 and 60, 200 kW system located on the Sears building in Chicago are presented.

ADVANTAGES OF SINGLE ANTENNA AND LINE

The advantages of common sites for multiple channels are easily recongnized. Real estate and tower costs are lower. The reduced competition for locations that satisfy the FAA and local governments provides faster and smoother approvals. All broadcasters benefit from the common orientation of receiving antennas.

For most broadcasters, the major consideration is economic. The use of common antenna and transmission line hardware further reduces the costs in comparison with multiple-antenna installations. The multichannel transmission system can even reduce the cost of the tower relative to mounting separate transmission systems on a common tower. Figure 1 compares a dual antenna installation and a multichannel antenna system on one tower. Even with the additional cost of the transmitter combiner, the savings using the broadband, multichannel system is significant. Although this example compares the case of two channels, the more substantial savings available by adding additional channels to the multichannel system is obvious.

There are some applications where a broadband antenna system is the only solution. Existing structures are often limited in available aperture for antenna mounting which prevents the use of multiple-antenna installations. The Sears channels 50 and 60 antenna, which is discussed later, is an example where only a multichannel antenna would fit the available space.



BUDGET COSTS

Dual Antenna		Dual Channel
610	Tower, 1200'	500
400	Antennas	250
300	Waveguide, WR1500	150
-	Combiners	35
1,310	Totals (k $)	935

Savings $ 375 k

FIG. 1 Cost comparison of dual-antenna vs. dual-channnel.

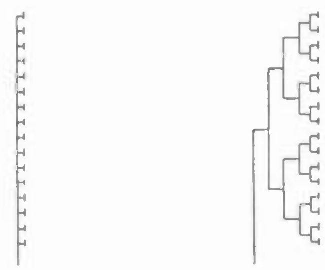
ANTENNA CONSIDERATIONS

Broadband vs. Narrowband

Multiple channels on a single antenna require the antenna to be wide band in both pattern and impedance (VSWR) characteristics. As a result, the antenna design represents a significant departure from the narrowband, single channel pole antennas that are commonly used for UHF. The typical single channel UHF antenna uses a series feed to the individual radiating elements while a broadband antenna has a branch feed arrangement. The two feed arrangements are shown schematically in Figure 2.

At the design frequency the series feed provides co-phased currents to its radiating elements. As the frequency varies the electrical length of the series line feed changes such that the radiating elements are no longer in-phase outside of the design channel. This electrical length change causes significant beam tilt out of band and an input VSWR that varies rapidly with frequency.

By contrast the branch feed configuration employs feed lines that are nominally of equal length. Therefore the phase relationships of the radiating elements are maintained over large frequency spans. This provides vertical patterns with stable beam tilts as required for multichannel applications. Broadband VSWR performance is also possible since the input impedance is essentially the average of all the radiating element impedances.



Series feed (narrowband) | Branch feed (broadband)

FIG. 2 Series feed vs. branch feed.

The basic building block of the multichannel antenna is the broadband panel radiator. The individual radiating elements within a panel are fed by a branch feeder system that provides the panel with a single input cable connection. These panels are then stacked vertically and arranged around a supporting spine or existing tower to produce the desired vertical and horizontal radiation patterns.

This design provides great flexibility for accommodating the multiple requirements of gain, power handling, and horizontal radiation pattern shape. For increased gain, stack more panels vertically. If greater power handling is required, use more panels either vertically and/or around the mounting structure. Since there is freedom to vary the phasing, power division, and pointing direction of the panels, numerous custom horizontal radiation patterns are possible.

It is this flexibility that makes it difficult to put absolute limits on the performance capabilities of multichannel antenna systems. The scope of this discussion is limited to the higher power systems which require waveguide for the power transmission from transmitter to antenna.

The ability to combine multiple channels in a single transmission system depends upon the bandwidth capabilities of the antenna and waveguide. The antenna must have the necessary bandwidth in both patterns and impedance (VSWR). It is possible to design UHF antenna systems for lower power applications using coaxial transmission lines that provide whole band capability. For the high power systems, the waveguide bandwidths set the limits of channel separation.

Pattern performance is not the limiting factor. As frequency increases the horizontal pattern circularity deteriorates, but this effect is generally acceptable. Also, the electrical aperture increases with frequency which decreases the vertical pattern beamwidth. If a high gain antenna was used over a wide bandwidth, the increase in electrical aperture might make the vertical pattern beamwidth unacceptably narrow. This is not a problem with the channel limits set by the waveguide.

For the antenna, it is primarily VSWR that limits its bandwidth. Broadband VSWR performance is accomplished by the following measures:

a) branch feed system
b) broadband individual panel VSWR
c) elements fed with phase pertubation scheme[1]
d) properly located discontinuities in feed system[1]

Item c), phase pertubation, generally occurs as a natural byproduct of providing null fill and beam tilt phasing over an antenna's vertical aperture. As a result the similar impedance characteristics of the panels cancel out at the antenna's input. It is also possible to incorporate phase perturbation within individual bays of an antenna to improve the impedance characteristics of the bay. By properly locating discontinuities in the feed system, the impedances of the multiplexed channels are matched and an improvement in overall bandwidth occurs.

By use of the above techniques, it is possible to offer an antenna input VSWR specification for each of the combined channels that assures excellent picture performance comparable to that achieved by single channel antennas. A typical VSWR specification is:

Antenna Input VSWR :	1.05:1	Visual carrier
(Each channel)	1.10:1	Visual upper sideband

Horizontal Pattern Capabilities

Due to the physical design of broadband panel antennas their cross-section is larger than the typical narrowband pole antenna. Therefore, as the operating frequencies approach the high end of the band, the circularity (average circle to min or max ratio) of the omnidirectional antennas generally deteriorates. For example, the economical 4 panels per bay antenna configuration has typical circularities of $\pm$ 2 dB at channel 26 which increases to $\pm$ 3 dB at channel 69.

It is illuminating to consider what meaningful effect this has on coverage. Figure 3 shows a 4 around horizontal pattern with $\pm$ 2.5 dB circularity and its RMS circle. Since an omnidirectional antenna's ERP is based on the RMS gain value, the Grade B coverage distance is actually greater in the maximum directions as well as less in the minimum directions. For this case, a variation of +2, -4.5 miles relative to the FCC calculated Grade B distance of 51.5 miles (1200 ft. hgt.; 2.5 MW ERP) is obtained. The total coverage area calculates to within 1.1 % of the RMS circle coverage area on a square mile basis. In essence the area of service is nearly the same as a perfectly omnidirectional antenna. If an area of higher population exists, the pattern maxima could be orientated in that direction.

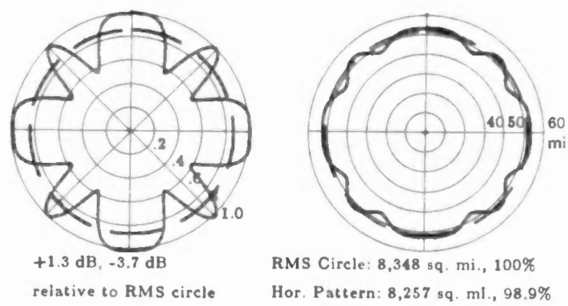


FIG. 3 Horizontal pattern vs. coverage.

Improved circularities are possible by arranging additional panels around the supporting structure. Previous installations in Canada and England have used 5, 6, and 8 panels per bay. These are illustrated in Figure 4 along with measured patterns at different operating channels. These approaches are often required for power handling considerations especially when three or four channels are involved.

The flexibility of the panel antenna allows directional patterns of unlimited variety. Some of the more common applications are shown in Figure 5. The peanut and cardiod types are often constructed on square support spines as indicated. Another type of cardiod is possible by side-mounting on existing triangular towers. The Sears channels 50 and 60 is an example.

Also indicated in Figure 6, is the capability to provide different horizontal radiation patterns for each channel. This is done by changing the power and/or phase to some of the panels in the antenna with frequency.

Most of the above antenna configurations are possible using a circularly polarized panel. This panel can also be adjusted for elliptical polarization with the vertical polarization receiving less than 50% of the power. Using a circularly polarized panel will reduce horizontally polarized ERP's by half.

A summary of the above mentioned antenna configurations is given in Table I. The pertinent performance characteristics for multichannel applications are indicated. These include circularity for omnidirectional cases, gain, input power, and total ERP. This is the ERP available for division among the channels operating on the antenna if maximum input power was applied. If circular or elliptical polarization is involved, the ERP allotted to one channel represents the sum of the horizontally and vertically polarized ERP's.

Table I is a partial list of the possible antenna configurations. It serves as a systems planning guide only. Desired channel combinations outside the recommended spacings still may be possible as was the case for Sears channels 50 and 60.

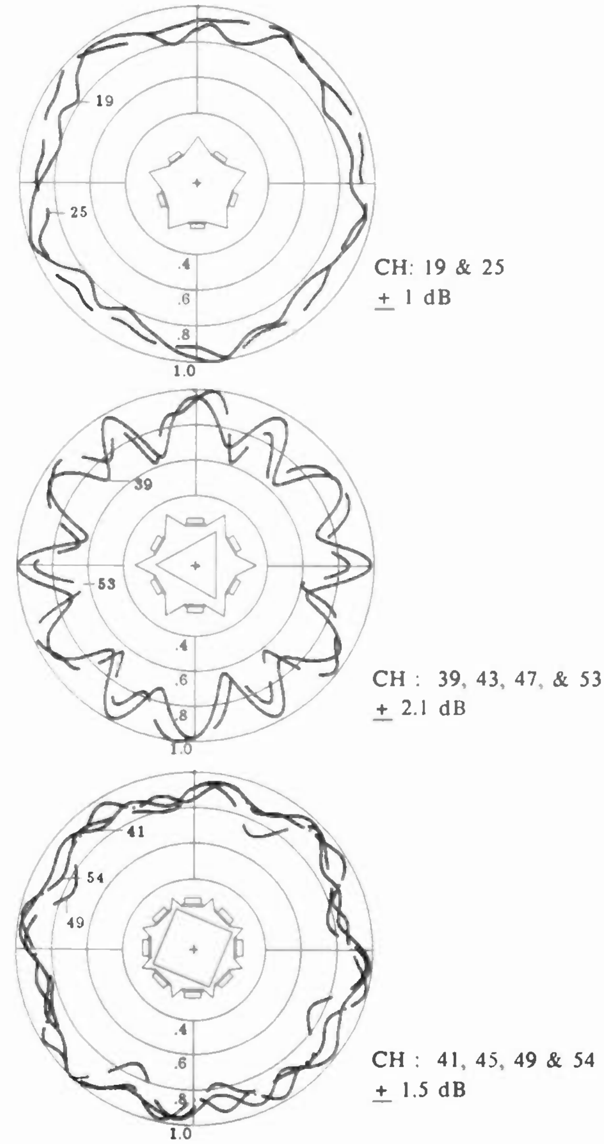


FIG. 4 Measured patterns; 5, 6, & 8 panels per bay.

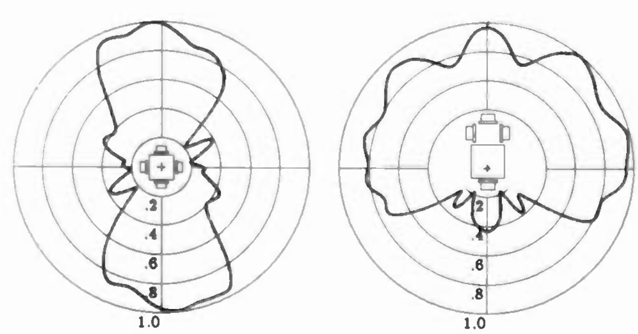


FIG. 5 Measured Peanut and Cardiod patterns.

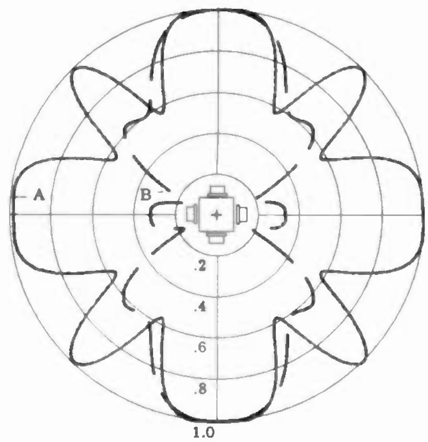


FIG. 6 Calculated patterns: CH "A" - omni
CH "B" - peanut

TABLE I

UHF MULTICHANNEL ANTENNA CONFIGURATIONS

OMNIDIRECTIONAL

ANTENNA/WG	RECOMM CHANNELS	TYPICAL CIRC (± dB)	MAXIMUM RMS GAIN (PWR)	MAX POWER (kW)	TOTAL ERP (kW)
4 PANELS per BAY					
WR-1800	14-39	2.0	31	230	7130
WR-1500	30-56	2.5	31	210	6510
WR-1150	52-69	3.0	31	195	6045
5 PANELS per BAY					
WR-1800	14-39	1.0	31	285	8835
WR-1500	30-56	1.5	31	265	8215
WR-1150	52-69	2.0	31	245	7595
6 PANELS per BAY					
WR-1800	14-39	1.5	31	345	10695
WR-1500	30-56	2.0	31	315	9765
WR-1150	52-69	2.5	31	290	8990
8 PANELS per BAY					
WR-1800	14-39	1.0	31	460	14260
WR-1500	30-56	1.5	31	420	13020
WR-1150	52-69	2.0	31	390	12090

DIRECTIONAL

ANTENNA/WG	RECOMM CHANNELS	TYPICAL HRP GAIN (PWR)	MAXIMUM PEAK GAIN (PWR)	MAX POWER (kW)	TOTAL ERP (kW)
PEANUT					
4 PANELS per BAY					
WR-1800	14-39	2.4	74	125	9250
WR-1500	30-56	2.4	74	115	8510
WR-1150	52-69	2.4	74	110	8140
CARDIOID					
3 PANELS per BAY					
WR-1800	14-39	1.8	56	170	9520
WR-1500	30-56	1.8	56	160	8960
WR-1150	52-69	1.8	56	145	8120
CARDIOID					
4 PANELS per BAY					
WR-1800	14-39	1.8	56	230	12880
WR-1500	30-56	1.8	56	210	11760
WR-1150	52-69	1.8	56	195	10920

NOTE: 1) Max Gain - highest gain typically offered to maintain stable performance.
2) Total ERP - available ERP for division among channels if maximum input power applied.
3) CP or Elliptical polarization - allotted ERP is sum of horizontal and vertical polarizations.

TRANSMISSION LINE AND COMPONENT CONSIDERATIONS

The choice of the proper size and type of transmission line and components must occur in the first stages of any multichannel system consideration. A properly chosen transmission line system must fulfill the requirements of; propogating all channels at the same time, operating at the total combined output transmitter power, offering minimal insertion loss and VSWR on all channels, and providing high isolation between the transmitters. Guidelines which summarize these requirements have been developed to aid in the selection of a proper system.

Frequency Selection, VSWR, Insertion Loss

The first things to consider when defining a system are the frequencies to be combined and the VSWR and insertion loss specifications desired. There are two basic limitations on the allowable frequencies of a multichannel operation; the minimum allowable channel separation needed for proper (-35dB min.) transmitter isolation by the combiner and secondly the largest channel separation at which the desired maximum VSWR and insertion loss for each channel can be obtained in the transmission line and antenna.

The first restriction is relatively constant across the UHF band for particular types of combiners. For the combiner developed by MCI, a spacing of approximately four channels is required between the combined channels to provide this minimum isolation. Different combining techniques might allow lower spacing if required for a particular situation, however, the VSWR and insertion loss may not be as low as that obtained in this example. The second, and more severe limitation on the frequencies to be combined in a multichannel system is the maximum VSWR and insertion loss allowable for each channel. The VSWR performance is highly dependent on the desired system layout, number of elbows, transitions and other components needed to carry the signal to the antenna. The maximum bandwidth and minimum residual VSWR associated with different components varies widely with their design and configuration in the system. Thus all components should be considered before a final maximum system bandwidth is determined. The restriction then is based on the frequencies allowed to propagate in the waveguide and associated components of the sytem.

Figure 7 shows the channels over which the three standard rectangular waveguide sizes can propogate effectively and also the channels which are usually specified for single channel operation. These waveguide sizes are standard, custom sizes are available for systems whose frequencies might fall between a standard size. From this figure the range of allowable channel combinations for each waveguide size can be seen. In this figure the total range does not consider any other component bandwidth restrictions and assumes a straight waveguide with no elbows or transitions.

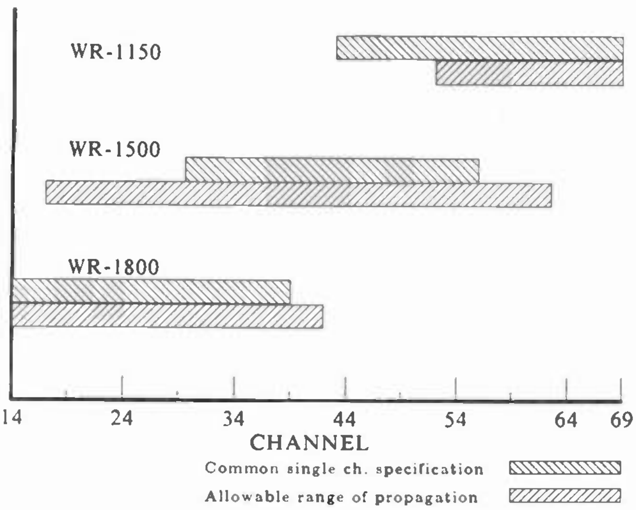


FIG. 7 Standard waveguide channel usage.

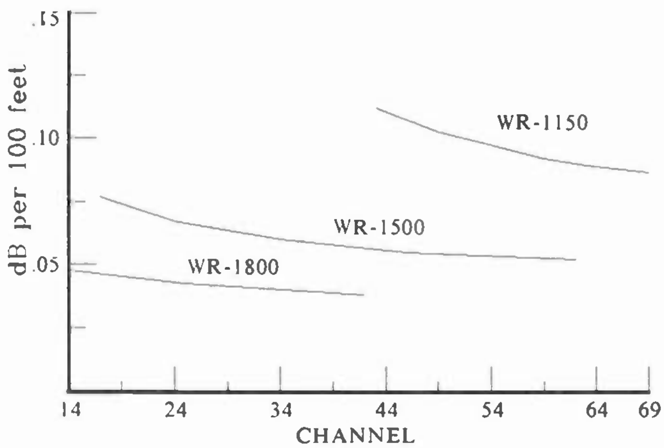


FIG. 8 Standard waveguide attenuation curves.

The question then is what performance can we expect for elbows, transitions and other components. A complete analysis of the bandwidth of each component in a prospective system is far beyond the scope of this presentation. Below we will generalize the performance of several important components. For our application the standard definition of bandwidth does not apply. Instead, bandwidth is considered the band over which satisfactory separate fixed tuning of each channel is possible and the channels in between are ignored. Obviously this type of tuning becomes difficult when large residual VSWR's must be tuned out on each channel. Standard waveguide transitions have been made which cover more than an entire waveguide band but have fairly large residual VSWR's(1.3:1) or do not handle large amounts of power. Optimized transitions have been designed to cover most of the waveguide band (30% bandwidth) with very good performance (<1.08:1 VSWR) at high power levels. Elbows present the other main restriction to VSWR performance. Standard single mitered elbows have untuned bandwidths (<1.08:1) of approximately 5-10% while optimized double mitered elbows can operate over a 30% bandwidth with similar performance. In summation, the total VSWR of a system can be determined after choosing the frequencies which can propogate in the waveguide and examining the number of elbows, transitions and other components which increase the systems VSWR and lower the maximum bandwidth available. If a system is designed with only one transition and one elbow a wider frequency range, nearing that of the waveguide, is available. If there are several transitions, elbows and bends then the range will be much more restricted.

Figure 8 gives the attenuation or insertion loss of these same standard waveguides over their frequencies of propogation. Although insertion loss should always be as low as possible, the restrictions placed on the choice of the waveguide size above limit the optimization of a system for insertion loss. If it is found that all channels can propogate in two different size waveguides then the size which offers lower insertion loss should be chosen. This would occur when the combined channels are spaced close together and are near the end of one size operating range and in the middle range of another.

Total System Output Power

The system must be able to handle the combined total power of both channels. Current waveguide and components have theoretical power handling capabilities of many megawatts although only tested up to 240 kW, the current available maximum transmitter power for a single UHF station. Higher combined power levels may be possible dependent on the necessary components and frequencies.

Waveguide Section Lengths

The lengths of waveguide sections in a long waveguide run must be calculated such that the reflections due to flange mismatches do not sum in phase causing high VSWR within any of the desired channels. This standard practice for waveguide installations becomes more complicated when more than one channel must be considered and may result in lengths which are not standard. A formula has been derived to aid in the proper selection of this length.

Combiner Considerations

A combiner must be designed for each installation which provides high isolation, low VSWR, and low insertion loss for each of the transmitters in the system. Two basic types of combiners are currently in use for multichannel operation. The branch type, where a separate filter is used for each input and combining occurs at a tee or common point, has the disadvantage of offering relatively low isolation between inputs and is much more sensitive to impedance changes of the source and load. The constant impedance type uses two hybrids and bandpass or band-reject filters tuned to one of the input frequencies. They offer high isolation, that of the hybrid plus that of the filter, excellent impedance characteristics, and relatively low loss.

Constant Impedance Combiners

Figure 9 shows the schematics for two types of constant impedance combiners. The choice of filtering in the combiner is dictated by the bandwidth needed (6 MHz for TV), the spacing of the input channels and the insertion loss desired.

In the band-reject combiner (CIBR), figure 9a, signals fed into the injected port are reflected by the apparent short circuit of the filter at this narrow or notch frequency and appear at the output port. Signals fed into the through port are split and allowed to pass unimpeded through the filters and also appear at the output port. This type, used in standard CIN visual/aural TV diplexers, performs well when the frequencies to be combined are closely spaced and of narrow bandwidth. To obtain high isolation over the full TV bandwidth a band-reject filter requires several cavity sections and an associated high insertion loss results for the through frequency.

In the bandpass combiner (CIBP), figure 9b, again signals fed into the injected port are reflected by the apparent short circuit of the filter and appear at the output port. Signals fed into the through port are split and now pass through the narrow passband of the filter and again appear at the output port. Thus the narrow ports of the two types of combiners are reversed, the injected port of the band reject and the through port of the bandpass are narrow. Bandpass filtering requires somewhat wider channel spacing but offers lower insertion loss and higher isolation along with better thermal stability than that of the band reject. Thus the benefits of bandpass combiners make their use attractive for this application.

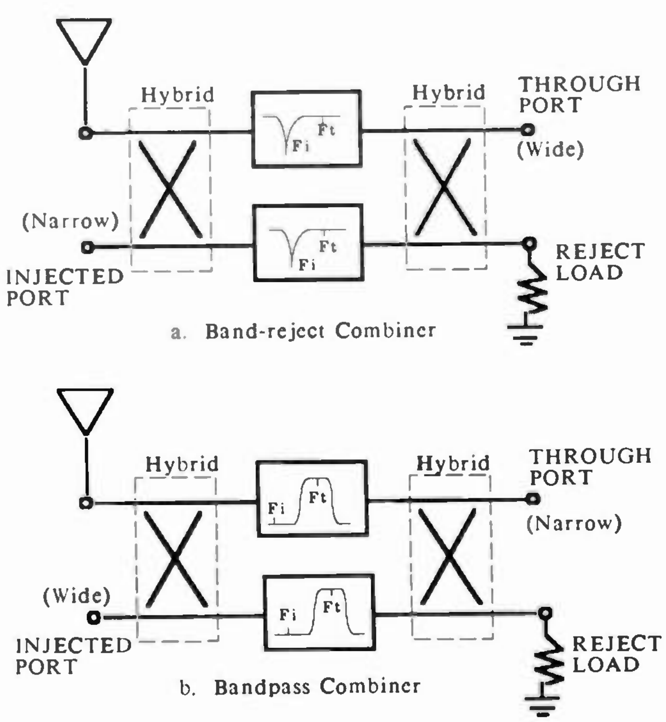


FIG. 9 CONSTANT IMPEDANCE COMBINERS

Combiner Performance Obtainable

The performance of the CIBP combiner can be predicted once the frequencies and the minimum isolation have been established. An isolation value of -35dB minimum is satisfactory to avoid problems of intermodulation. As stated previously four channels of space are required to ensure this isolation with the current design. A VSWR of less than 1.1:1 is obtainable for the CIBP combiner on both channels, the through port having a somewhat higher VSWR than the injected port because of travel through the filter. Insertion loss for the through port will again be somewhat higher than the injected port because of travel through the filter. This difference is minimized by design optimization of the bandpass filters for the frequencies involved. Insertion loss of less than 0.25dB for the through channel and 0.1dB for the injected channel are obtainable.

SEARS CHANNELS 50 & 60 SYSTEM

Antenna

On the Sears Building west tower in Chicago, 60 feet of aperture was available to install an antenna for the two channels. The supporting structure is a 5.5 foot face triangular tower enclosed by an 8 foot diameter radome. The space limitations were accomodated by mounting 8 bays of 4 panels per bay around the appropriate leg. The biggest space difficulty was locating the WR-1500 waveguide feed and power divider. They were located on the outside of the 5.5 foot tower section on the face that was opposite the antenna leg.

A cardiod horizontal pattern to cover up and down the Lake Michigan shore line and inland over Chicago and suburbs was provided. The system was designed to provide each channel a peak ERP of 5 MW with a nominal 100 kW input. All antenna design specifications were met or exceeded.

SEARS ANTENNA SPECIFICATIONS

Aperture	48.0	ft
Peak Gain	53.7	pwr/dipole
Beam Tilt	1.0	dgs
Null Fill, first	20.	percent
second	10.	percent
Power Rating (10% aural)	200.	kW
VSWR	1.05:1	visual carrier
	1.10:1	across channel

Figure 10 shows the antenna configuration in plan view on the tower and the horizontal pattern. The one bay pattern measured on a tower section overlays the calculated free space pattern of the panels without tower. Since the tower structure was already in place, it was not optimum for the application. The tower blocks radiation behind the antenna as expected. However, the measured pattern still provided an excellent match to the geographical coverage requirements.

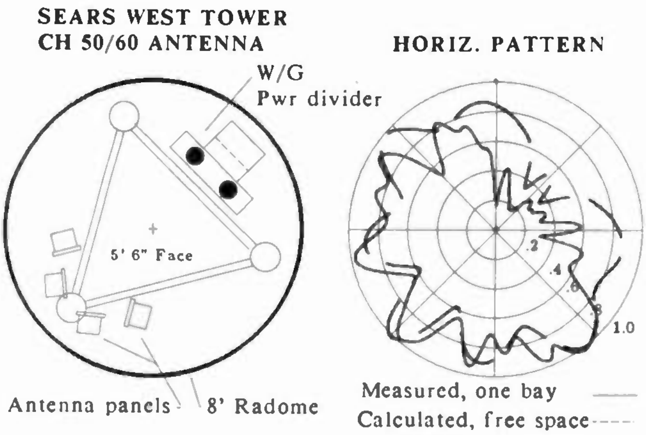


FIG. 10 SEARS 50/60 plan view and horizontal pattern.

Waveguide

The combiner and feedline used standard WR1500 rectangular waveguide and components. Figure 11 shows the components used in the installation including various E and H plane elbows, a step twist, three offsets and a top mounted waveguide power divider with four coaxial outputs to the antenna system. A length of 141.75 inches was chosen as the optimum length for waveguide sections. Figure 12 shows the frequencies where flange reflections occur for this length. Special transitions were designed to provide low VSWR on both channels (9% bandwidth) for accurate testing. A new five port switch with a much larger bandwidth than those currently available was designed and tested.

Excellent VSWR and insertion loss performance was obtained for this system. Figures 13 and 14 show the final VSWR obtained for the transmission line system and antenna ahead of the combiner on both channels. Figures 15 and 16 contain the final VSWR, insertion loss, and isolation data obtained for the combiner on both channels. Thermal stability tests were run on the bandpass filters for channel 50 over a 60 degree Fahrenheit range with very little change in the VSWR or insertion loss. Both stations are currently in full power operation at 100 kW each for a total of 200 kW in the line and antenna.

Conclusion

Multichannel installations have been common at FM and VHF TV for many years now with excellent results. Multichannel operation has now been introduced to high power UHF TV with similar results. We have shown the benefits of multichannel operation, and discussed antenna, transmission line, and component considerations for a proper multichannel system design. Data was presented for a representative 200 kW channel 50/60 system now in operation on the Sears Tower.

Multichannel system operation using waveguide is possible with current designs for stations which fall within approximately a 30% frequency bandwidth and are spaced more than 4 channels apart.

ACKNOWLEDGMENT

The authors wish to thank M. B. Anders and his staff at the Alan Dick & Company Limited for the information pertaining to broadband panel and Sears 50/60 antenna performance.

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1. M. B. Anders, "A Case for the Use of Multi Channel Broadband Antenna Systems", NAB, 1985.

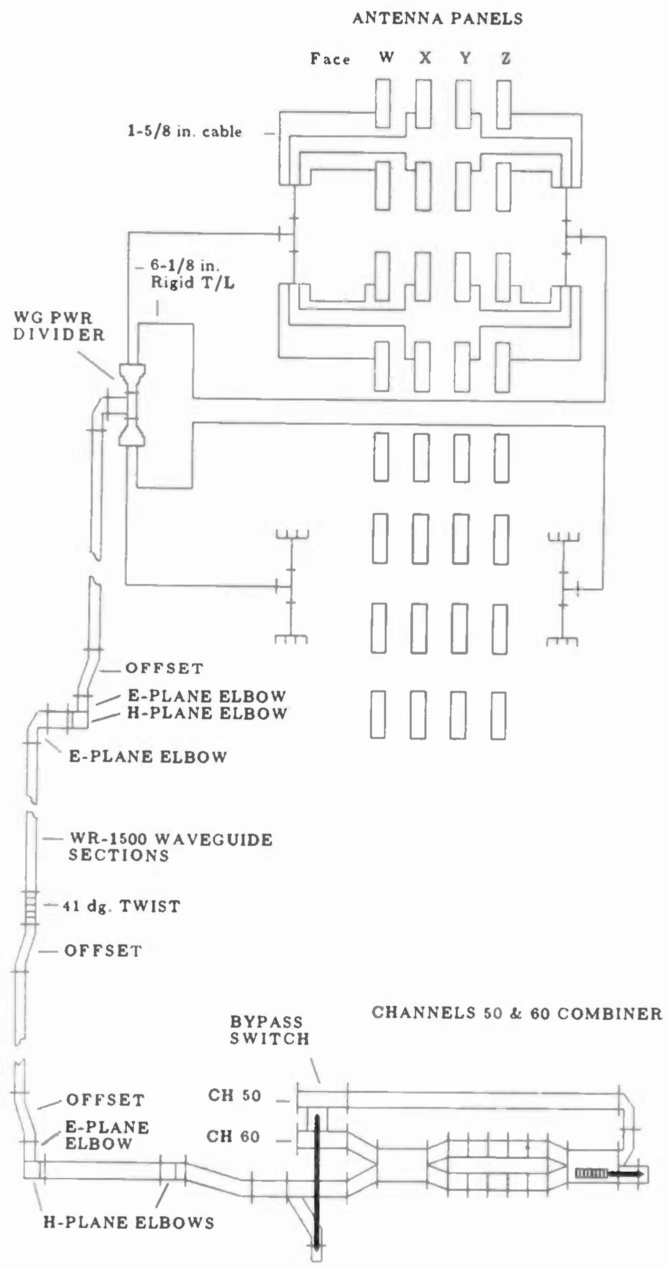

FIG. 11 SEARS CH 50/60 System Diagram

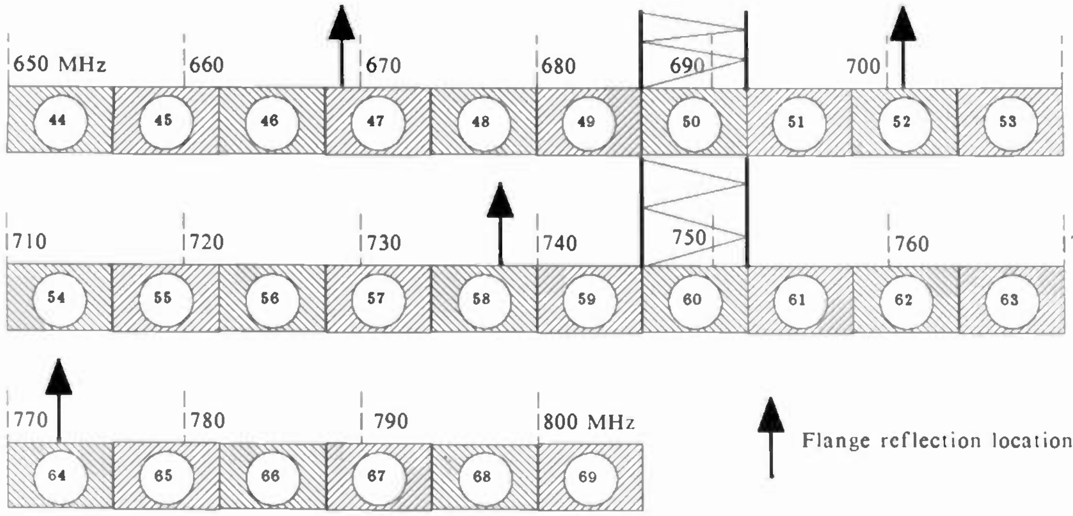

FIG. 12 Frequencies of flange reflections for 11' 9.75" WR-1500 waveguide line sections.

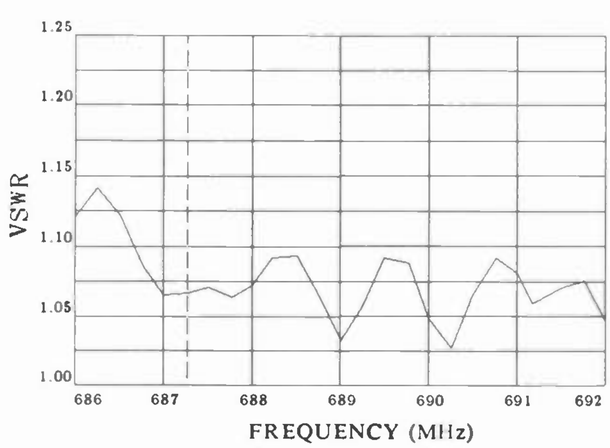

FIG. 13 Final System VSWR, channel 50.

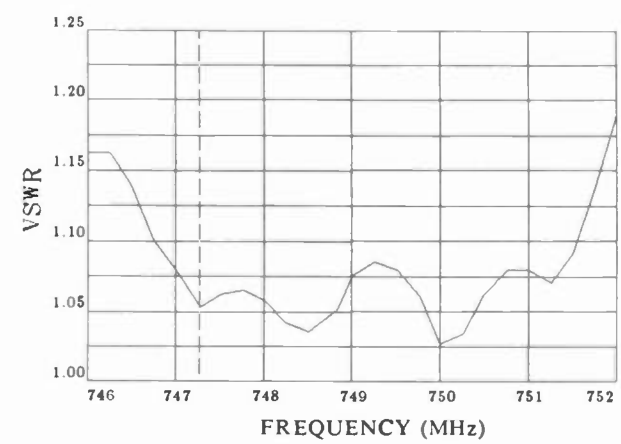

FIG. 14 Final System VSWR, channel 60.

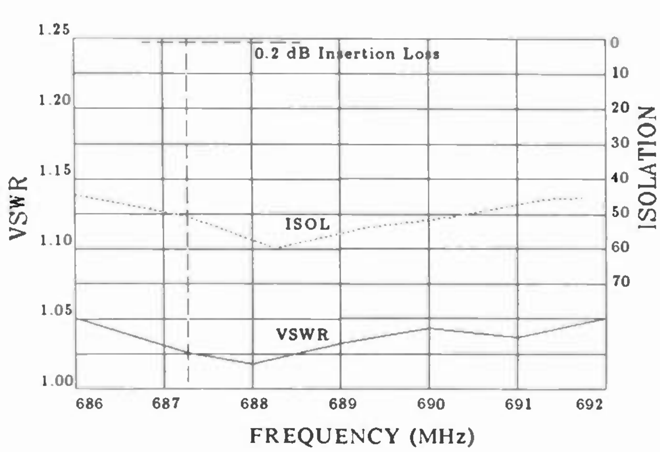

FIG. 15 Combiner VSWR, Isolation & Insertion Loss. Channel 50

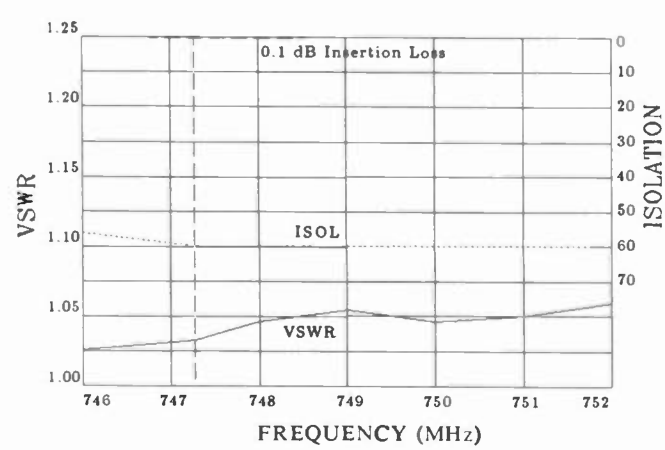

FIG. 16 Combiner VSWR, Isolation & Insertion Loss. Channel 60

DIGITAL TECHNIQUES FOR TELEVISION ANTENNA IMPEDANCE MEASUREMENTS

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One of the major equipment manufacturers has recently introduced a new digital device which essentially represents a new generation of network analyzers. This particular equipment will permit antenna measurements to be performed with a new degree of ease along with increased accuracy. One of the primary features of this equipment is that, it is self-contained in one single device without the need for external converters or generators. A complete automated test system can be assembled and transported to the station site easily and conveniently.

In the past, television antenna and transmission system measurements have been performed by network analyzers primarily using swept frequency techniques. Those methods utilized a sweep generator and a tracking receiver with directional couplers or reflection bridges to measure the return loss of the signal. The basic problem with this technique was that the system only indicated the return loss measured at the actual test equipment port. It was not possible using only these devices to determine the actual return loss or VSWR of an antenna without the effects of the transmission unless

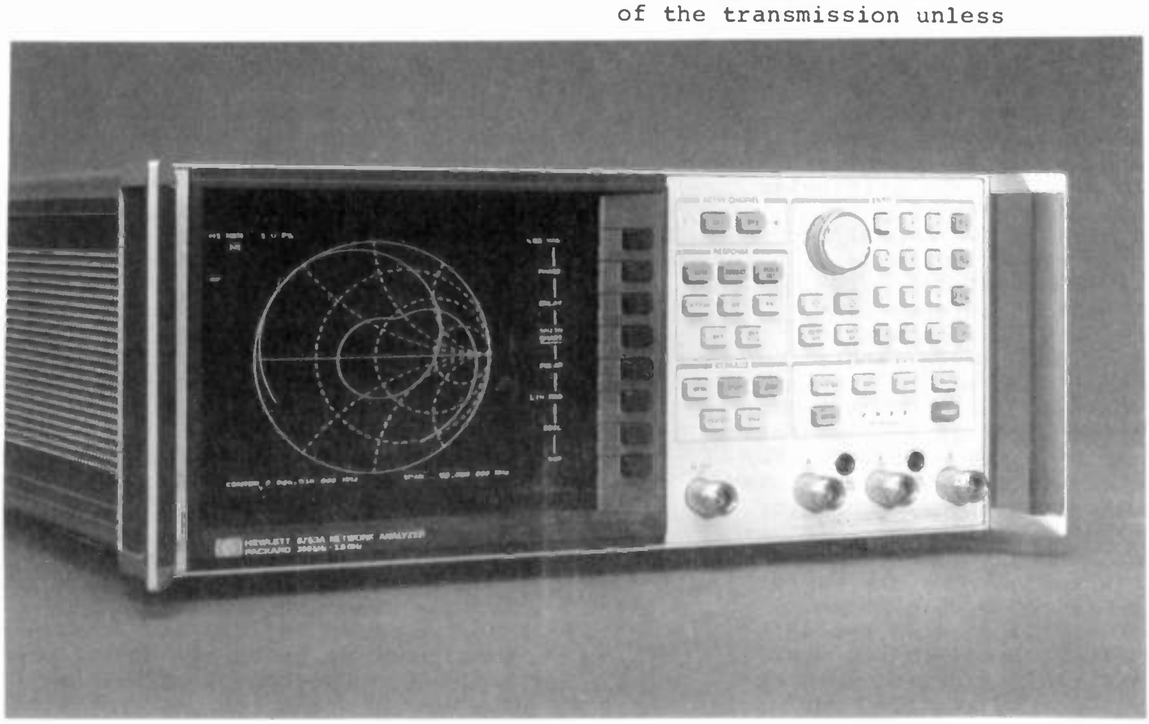

Figure 1
Hewlett-Packard Model 8753A Network Analyzer

the equipment was actually taken to the antenna input. Even when the equipment was taken to the antenna input, the display was normally only a representation on a CRT display. With some effort, it was possible using external devices to obtain a plotter representation of the return loss of the system.

generator,the computer to operate all of this equipment and the plotter. Obviously, this required taking multiple pieces of equipment to the site.

A completely automated system of this type could be used to attempt to eliminate the effects of the transmission

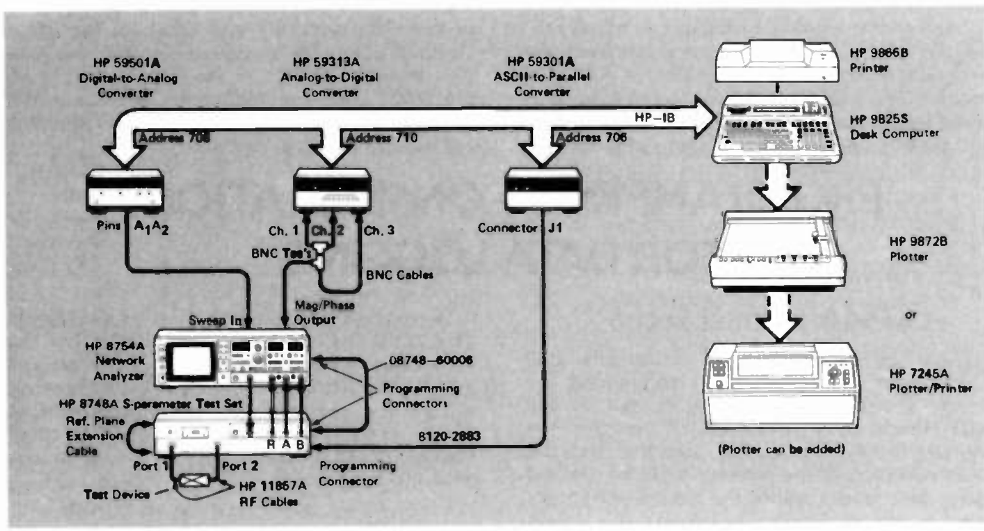

Figure 2
Previous Method of performing automated measurements

A complete automated system using previously available equipment is shown in Figure #3. In this particular example, A Hewlett-Packard Model 8754A Network Analyzer is shown connected to an S parameter test set. In the broadcast antenna configuration, this network analyzer might more likely be found connected to a return loss bridge. However, the test equipment still is representative of what might be expected. For extremely accurate measurements, a synthesized signal generator was usually included. This was primarily due to the fact that the earlier network analyzers exhibited too high a degree of incidental FM when operating at a single frequency to provide the extremely accurate measurements normally required for television antenna systems. Therefore, the overal system would consist of a network analyzer with a return loss bridge, at least one digital to analog converter as well as a synthesized signal

line from the overall measurement results. This was done by connecting the system to a known load and determing the characteristics of that load. The load characteristics were recorded digitally in the computer at discreet increments of frequency across the band being measured. The load was then transferred to the far end of the transmission line system and the measuring equipment connected to the input of the transmission line. Measurements were then performed at each discreet frequency at the input of the transmission line and the effects of the load were mathematically removed from the measured return loss at each of the discreet frequencies. This then was used to identify what were believed to be the characteristics of the transmission line. The system would then be completed by connecting the transmission line to the antenna and a third set of measurements would be performed. The effects of the transmission line would then be mathematically removed from this newest set of measurements with the result being a fair representation of the antenna itself.

While certainly a state-of-the-art technique at the time this equipment first became available, there were certain inherent problems. First, measurements were being performed of variables which were very close to the noise floor of all pieces of equipment. Therefore, the equipment was being asked to work at essentially its maximum range with this data being used to mathematically evaluate more measurements being taken at basicallythe maximum range of the equipment. The chances for error are obvious. Second, the correction constants being developed wre simple and only concerned the actual phase and magnitude of return loss at each discreet frequency. Finally, the largest problem with the entire system was the bulk of system which had to be transported to the site.

Hewlett-Packard, as well as other manufacturers developed more sophisticated systems which included self-calibrating or enhancement programs to eliminate far more of the errors involved. Unfortunately, these systems were considerably higher in price, as well as being even bulkier. The ability to take these test equipment systems out to a remote antenna site was questionable as would be the common-sense of an engineer who would drag $100,000.00 worth of equipment out of the laboratory.

The 8753A, which block diagram is shown in Figure 3, is a new system which does not really function in the manner of the previous portable network analyzers. The 8753A utilizes digital techniques to permit antenna measurements to be performed in a very simple, straight-forward manner. The device does not actually use a swept frequency technique in the traditional form. Instead, a synthsized generator is stepped across the band of interest in discreet units. This can vary from as small as three test points over the interval to 1601 test points. The default number is 201 points which appears to give a very smooth appearing display. The device is also capable of multiple self-calibration techniques to essentially remove the errors that are generated by the transmission lines,`interconnecting cables and hardware. This system consist of a digital synthesized signal generator which is controlled by the main processor and software. The return signal used is phased locked to the three receiver systems of the generator. The ouputs of these receivers are then processed in first a digital I.F. mutiplexer system then converted to digital data to be processed by the main and fast processors themselves.

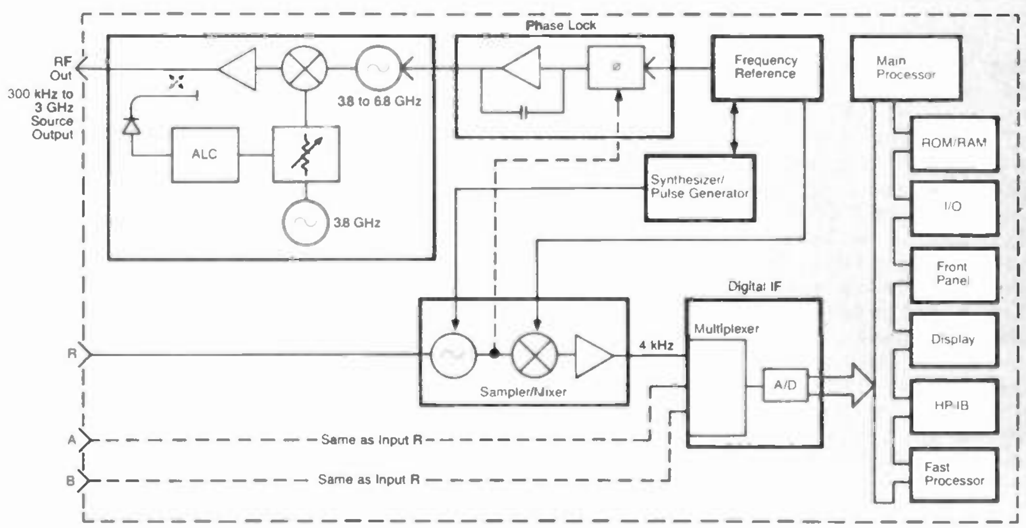

Figure 3
Model 8753A Block Diagram

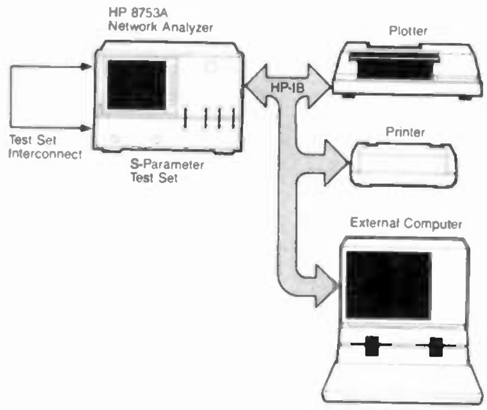

Figure 4
Automated Measurement System
Using the 8753A

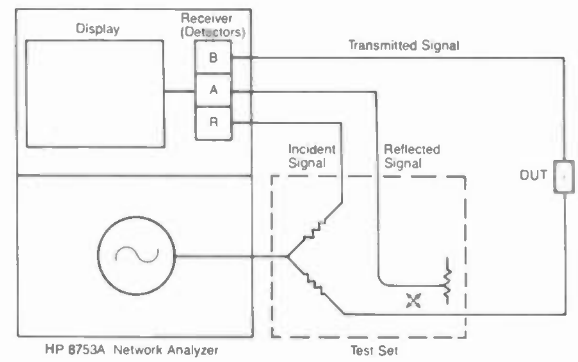

Figure 5
Test Setup for Digital Measurements

A complete automated test system using the 8754 is shown in Figure 4. This represents a more sophisticated system than was previously shown for the earlier equipment and associated extra devices. Completely automated measurements are performed by using an external computer which may be as small as an H-P 41C to direct the measurements of the network analyzer itself. The outputs of the network analyzer are shown on its built-in CRT display. The results can also be printed on an attached plotter or printer, or it may be stored in the computer to be plotted at a later date. The greatly reduced number of pieces of equipment as well as the reduced price permit antenna measurements to be performed on site in a much more convenient and economically feasible manner.

The device is totally software driven. Even the intensity of the presentation on the CRT is adjusted digitally by the internal program. Return loss of transmission parameters are displayed as log magnitude, linear magnitude, polar, Smith Chart or VSWR displays. Multiple markers can be assigned to identify various parameters such as the bottom and top end of the television channel, visual and aural carriers as well as having one remaining marker which is adjusted to any frequency of interest. Digital Gating allows all parts of the display or information outside the band interest to be eliminated.

Figure 5 shows the simple test setup required for digital measurements of an antenna system. In this particular case, a test set is shown providing both reflected signals and through signals from a device under test. This might be a piece of transmission line or a television diplexer. In measuring the antenna itself, obviously, the transmitted signal is not available for use as an input to the test equipment.

One of the primary features of this device which makes it so usable in the broadcast industry is a self-enhancement or calibration system that is built-in to the equipment itself. A single port calibration test can be performed either at the output of the direct reflection test set or at the end of a lengthy piece of transmission line. This built-in claibration test utilizes open, short and calibrated load terminations. This calibration corrects for the magnitude and phase of frequency response errors as well as compensating for the directivity of the ports and source match itself. In addition, a time delay to a distant calibration point is automatically entered by the device.

To explain what this really means, let is be assumed that it is being attempted in the field to measure a faulty television antenna which is installed at the top of the tower. The test equipment can be connected to the transmission line system at the patch panel in the transmitter room. First, a set of calibrations might be performed on the equipment itself with the open, short and load terminations applied to the test cables at the point where they would be connected to the adapters to the larger coaxial cable.

This results in a set of correction parameters being generated internally in the machine which will be accurate as far as presenting the impedance or reflection data as measured at the patch panel. This set of data can then be stored to be recalled at a later point in the overall measurement process without having to recalibrate the system a second time. The short and terminations can then be taken to the top of the tower and be connected to the far end of the transmission line. The system can be calibrated as of that location simply by placing a short on the adapter, attaching a shielded open circuit and then attaching a calibrated termination. When the system is now connected back to the antenna, the actual display on the CRT will be the input measurements being performed albeit the transmission line affects. This is not only eliminating the affects of the transmission line system itsel, but also eliminating any errors which are introduced into the overall measurement process by interconnecting cables, adapters and directional coupler of reflection test set itself.

It is possible to display both log magnitude and Smith Chart representations on the CRT at the same time. The scales of these presentations can be adjusted individually and stored when needed. The device will copy the CRT presentation directly to an XY plotter without the need for external computer interface equipment. It also can be recorded for further analysis in the office. When being usedin the polar display, the device presents the actual physical components which are equivalent to that load impedance. That is to say, that it reads out the actual resistance and inductance or capacitance values that represent a series circuit having the impedance characteristics shown.

With digital processing, this device is capable of working in a fairly hostile enviroment. In the presence of R.F. interference, digital smoothing or averaging is used to eliminate those parameters which keep varying in a random fashion. That is to say, external R.F. is eliminated by digital averaging process. It is possible to conduct fairly extensive measurements in the presence of a significant degree of unwanted R.F. signal. The system does fall apart when the R.F. signals on the trans-

Measurement Port Characteristics

Residual Error	before accuracy enhancement (1.3/3.0 GHz) 7 mm, 3.5 mm, type N		after accuracy enhancement (1.3/3.0 GHz) 7 mm	type N	3.5 mm
Directivity	35/30 dB		50 dB	44 dB	40/35 dB
Source Match	20/16 dB		40 dB	30 dB	30 dB
Load Match	20/16 dB		40 dB	30 dB	30 dB
Frequency Response[3]					
Reflection	±1.0 dB		±.05 dB	±.10 dB	±.10 dB
Transmission	±1.0 dB		±.03 dB	±.08 dB	±.08 dB
Crosstalk	90 dB		100 dB	100 dB	100 dB
	50 ohm systems	75 ohm systems			
Dynamic Range	100 dB	90 dB			

Figure 6
Results of built-in
Enhancement/Calibration Test

Figure 6 shows the results of the built-in enhancement or calibration tests. As you can see, reflection measurements are usually accurate to ± 1 dB with Type N connectors. After the test has been performed, these same test are now accurate to ±0.1 dB, an increase in accuracy of a factor of 10. In addition, the dynamic range of all of these measurements themselves is 100 dB for 50 ohm cables.

mission line are so great that the tracking receivers loose their phase lock. The first response of the device to high unwasnted R.F. levels on the transmission line is to attempt to reduce its output power to eliminate the R.F. signal. The operator is advised of this condition by a display printed on the CRT. When this does not work or when the phase-lock becomes lost on the receivers, the operator is advised that the measurement data is unusable by a message on the CRT. In cases of extreme R.F. interference, it will be necessary to use external amplifiers and directional couplers to drive the transmission line system. The return signal would then be reduced by a pad equal to the gain of the amplifier. Using the built-in calibration techniques, all of the responses and errors generated by this external can be eliminated from the overall measurement process maintaining essentially the same overall accuracy of measurements.

One feature which is available as an option for the 8753A is a built-in time domain reflectometer system. This does not operate as a true time domain reflectometer, but rather uses a Fourier transform technique to convert the frequency domain representaion to a time domain representation. It was first thought that this might be a very valuable option to provide R.F. pulse measurement signals to measure signals to measure antennas in that fashion. It was found that the relatively high Q of television and FM antennas eliminate the usefullness of this particular option. While extremely accurate for determining the fault in coaxial transmission lines themselves, the line would have to be terminated in a load other than the antenna for the measurement results to be at all useful. When the band-width of the device is limited sufficiently to keep it in the frequency range for the antenna, the accuracy of the fault location becomes unacceptable.

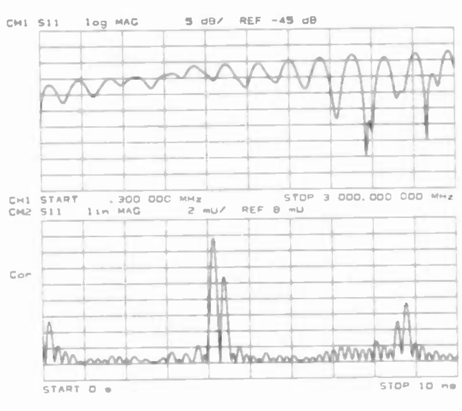

Figure 7
Two-Channel Display
Depicting Log/Lin as Magnitude

Figure 7 shows a two channel display with simultaneous presentations of both log magnitude and lin magnitude responses. In the same fashion, it is possible to replace one of these responses with a Smith Chart or Polar or VSWR display.

In summary, the new 8753A permits a new degree of accuracy to be maintained on in-field measurements without the necessity of external synthesized signal generators, analog to digital or digital to analog converters. All equipment is contained in one device which is small enough to be readily portable to remote antenna sites. Completely automated measurements are possible through the use of only a single small computer which as previously stated can be as small as the H-P 41C Although, somewhat larger ones are recommended for speed of measurements.

The author would like to thank Hewlett-Packard for their assistance in preparing this paper and for their permission to use various graphs and drawings from their technical material.

NATIONALLY COMPATIBLE, DS3 RATE (45 MB/S), CUSTOMER CONTROLLABLE, MULTIPOINT NETWORKS FOR BROADCAST TELEVISION DISTRIBUTION: POSSIBILITIES AND CHALLENGES

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ABSTRACT

This paper provides an overview of Bellcore's activities in the area of nationally compatible, customer controllable, DS3 rate (45 Mb/s), multipoint, digital networks for distribution to affiliates of broadcast quality NTSC television on terrestrial digital fiber optic and microwave interoffice transmission facilities (more than 215 cities are now interconnected) in a manner which rivals today's domestic satellite networks in cost, performance, reliability, flexibility and security. It also outlines a two year work program leading toward a multi-carrier, multi-supplier, multi-broadcaster trial of such a network, built to a yet to be specified standard, with disclosure of compatibility details and an open architecture for customized software for controllers. All are invited to participate.

Why DS3 Rate?

The local exchange carriers (LEC's) and the interexchange carriers (IEC's) have installed a large base of digital fiber optic and microwave transmission systems.[1-3] For example, AT&T recently announced that 215 cities are now interconnected digitally and US Sprint has announced plans for Canadian crossings to Ottawa and Montreal.

It is becoming increasingly possible to obtain DS3 rate (i.e., 44.736 Mb/s) digital channels from one place in the country to any other place by obtaining DS3 channels from the various common carriers and requesting that the carriers connect the various segments together.

Current fiber optic systems (565 Mb/s) carry as many as 12 two-way DS3's on a pair of fibers; current digital microwave systems carry two (4 Ghz systems) or three (6 and 11 Ghz systems) two-way DS3's on a single 20 or 30 MHz radio channel pair, respectively (analog radio systems provide only one video per radio channel). Single two-way DS3 per 40 MHz radio channel pair, 18 GHz systems are now available for use within cities for portable microwave pick-ups.

For interoffice transmission systems, the DS3 rate channel is the single common denominator in the United States and Canada even though there are a multiplicity of diverse suppliers of fiber and microwave transmission systems and a multiplicity of common carriers. The DS3 rate channel, which was established as a standard in the Bell System long before its break-up in 1984, has continued to this day so that essentially all embedded and future interoffice transmission plant can be interconnected at DS3 rate.

Although it is technically possible to assign more than one DS3 to the same service to effectively obtain a bit-rate of 90 Mb/s (two DS3's in parallel) or 135 Mb/s (three DS3's in parallel), the administrative difficulties of co-routing multiple DS3's either initially or when rerouting around catastrophic failures would be onerous.

Internationally, transatlantic digital fiber optic cables have been announced. There appears to be the potential for future United States-sourced NTSC video transmission to travelling Americans and conversely for European-sourced NTSC video for United States consumption. If this comes to pass, these too could be digitally interconnected to the domestic digital network at DS3 rate.

Spare Channels and Route Diversity

The interoffice transmission plant is engineered for high performance, (error rates typically better than 10^{-9} under normal conditions) and is generally backed-up by spare "protection" channels on a 1 by N (N = 1 to up to about 15 for fiber systems and N=1 to up to 11 for microwave systems) which are automatically substituted for channels whose performance degrades below about 10^{-6} in error performance. On radio channels, the switchover is hitless.

Within the continental United States, an intricate web of digital facilities is being installed which provides alternate physical route protection in the event of a catastrophic failure on an individual route (such as a cable cut or loss of a microwave tower).

Security of Terrestrial Networks

Satellites are vulnerable to launching and relaunching delays, double illumination (either intentional or unintentional), terrestrial interference, and theft of service (leading to scrambling). On the contrary, terrestrial digital networks are more likely to be available when needed, interference free and not vulnerable to theft of service (no picture scrambling required).

Performance of Current DS3 Rate Coders

Several suppliers have developed or demonstrated broadcast quality DS3 rate video coders.[4-9] One has been put into point-to-point commercial service by ABC for contribution with very satisfactory results. While all work over standard DS3 rate channels, no two suppliers' products are compatible with each other. The picture quality is excellent since compression to 45 Mb/s is relatively mild. Uncompressed pulse code modulation (PCM) would require a minimum sampling rate of 8.4 MHz for a 4.2 MHz bandwidth, and 8 or 9 bits per sample resulting in an uncompressed bit-rate of 67.2 to 75.6 MHz for video.

Various forms of differential PCM are used with varying degrees of complexity. All use intra-frame prediction. The simplest uses only one predictor (a preceding sample in the same line of the picture); more complex versions use as many as four different predictors (obtained from the same line, a previous line in the same field, and two previous lines in the previous field). The best predictor is automatically selected depending upon the changing (in time or in space) picture content and these more complex coders result in exceptional picture quality with no visible impairments with ordinary difficult motion sequences. The only known exception is with moving "zone plate" test patterns, which stress the predictors in both time and space, and with very fine black on white circular patterns.

Coders with 8 or 9 bit analog-to-digital converters, sampling rates of precisely 3 times the 3.58 MHz color subcarrier (i.e., 10.7 MHz) and 3 or 4 adaptive predictors have been shown to produce very high quality television pictures at DS3 rate. These coders provide full VITS and horizontal sync pulse transparency. Dependable, stable, signal-to-noise ratios (S/N) of better than 60 dB are obtainable regardless of distance; this is considerably better than the 54 dB required for end-to-end long distance terrestrial connections in EIA standard RS 250 B and also considerably better than that same standard requires for one-hop satellite links (56 dB). Clearly, the higher quality DS3 rate coders are suitable for distributing television signals from a broadcaster's master station to its remote affiliates.

Whether DS3 rate is also good enough for the so-called "contribution" phase, where S/N greater than 67 dB has been the norm, is still a bone of contention since the 67 dB requirement is an analog relic that recognized that short pieces had to be exceptionally good since long pieces would be worse due to the accumulation of analog degradations with distance. The critical question appears to be whether or not a DS3 rate coded picture can be decoded and further processed (e.g., chroma keying) acceptably - which, from discussions with ABC and CBS, has been shown to be the case-not whether or not 67 dB can be obtained.

DS3 Rate Coder Trends

There has been a tendency on the part of suppliers to develop different grades (high, medium and low) of DS3 rate coders for different applications (broadcast, CATV and teleconferencing). Over the last few years, the price and size of the highest quality has come down to the point where it is interesting to speculate that perhaps in the near future a high grade coder could be used for all applications with no significant cost penalty. This would also alleviate the carrier's burden of administering different grades of coders in a high churning environment and reduce inventory (less spares would be needed) problems. The larger combined demand for mixed applications should help to further reduce per-coder costs without sacrificing quality for any application.

To paraphrase a famous saying "a 45 Mb/s channel is a terrible thing to waste.". So is the human eye. So let us aim for high video quality in all situations and for interchangeability of coders between applications.

Bellcore's Technical Advisory (TA-TSY-000195)

A Technical Advisory issued by Bellcore in December 1984 to the carrier, supplier, broadcaster community, developed performance requirements for DS3 rate coders that permitted meeting RS 250 B end-to-end requirements with up to three digital encodings in tandem and allowed for a short haul analog piece, associated with each digital section.[10] For example, for signal-to-noise ratio, the overall 54 dB requirement was divided into three equal parts (58.8 dB) for access-interexchange-access and then further tightened to 60 dB to allow for a 67 dB analog loop in the event that interoffice digital facilities are available but only analog facilities are available in the loop plant.

In addition, the document calls for VITS and horizontal sync pulse transparency, 9-bit encoding rather than 8-bit to allow for multiple links in tandem, 3 f_{sc} sampling-locked because that has been demonstrated to provide excellent subjective results, forward error correction, a DS1 rate (1.5 Mb/s) embedded channel for multi-channel-television-sound (MTS)[11-12], and error mitigation. The document also specifies a method for objectively measuring signal-to-quantizing noise ratio with a low frequency sine wave superimposed on a variable level flat field and then notching out the sine wave to measure weighted quantizing noise.

Performance of End-to-End Digital Connections

Three coder/decoders in tandem, meeting the requirements of TA 195, would perform as well as current terrestrial analog facilities 4000 miles in length (S/N 54 dB). However, with a standard algorithm, all coders/decoders would be end-to-end compatible and therefore back-to-back codings and decodings could be eliminated at interfaces between LEC's and IEC's/IEC's, and the resulting end-to-end performance would be considerably better (S/N better than 60 dB instead of 54 dB).

Alternatively, a cross country end-to-end digital link with a DS3 rate coder (S/N better than 60 dB) would be superior to an analog satellite link that just met RS-250-B satellite requirements (S/N better than 56 dB).

Furthermore, if a DS3 rate digital tape recorder were available, delayed broadcasting for time-of-day differences could be accomplished without decoding and re-encoding. In this way the quality of a television program originating on the east coast and delayed for delivery on the west coast would be identical; in today's analog environment, the west coast is inferior by one analog taping generation to the east coast in such a situation. DS3 rate tape recorders are feasible since much higher bit rates are now tape recorded.

Standards Activities

The Exchange Carrier Standards Association sponsors the T1 Committee of the American National Standards Institute. Two Subcommittees, T1Q1.5 and T1Y1.1, are concerned with analog performance and a standard algorithm for DS3 rate coders. These committees include representatives from carriers (LEC's and IEC's), suppliers and broadcasters as well as Bellcore. To date, a joint ad hoc group has conducted subjective tests[6] and objective tests[7] for existing 45 Mb/s coders. The committee has invited all interested parties, both domestically and in Canada, Europe and Japan, to submit proposals for a standard DS3 rate algorithm. Submissions are expected in 2Q87.

Networking Possibilities

The extensiveness of the terrestrial digital fiber optic and microwave transmission networks available from the LEC's and IEC's at DS3 rate, offers the opportunity to create extensive customer controllable, digital multipoint networks rivaling the satellite networks in flexibility, cost, performance reliability and security. If a 2-way DS3 multipoint unit were available, a broadcaster could order up a tree-like multipoint network to connect a master source to any number of remote affiliates (typically about 200).

A study has shown that a multipoint unit with one input port and up to four output ports would be suitable for the large majority of applications.

With such an arrangement, the broadcaster could send control signals within the DS3 bit stream (without disturbing the video signal) to select which remotes should be receiving (Figure 1), to simultaneously allow a remote to send to the master, or through a loop around/through arrangement to send to the master and to the selected affiliates (Figure 2). If for any reason the master lost control, another remote could be allowed to take over control as a back-up for the master controller (Figure 3). In addition, on a polling basis, the 2-way control channel embedded within the DS3 stream could be used by the multipoint units and coders to send alarm and status information to the master station as well as a far-end-block-error indication for in-service digital error performance monitoring.

Furthermore, the control channel can be used to sectionalize troubles through a sequence of addressable loop-back/through commands that would allow digital loop-backs on individual ports on selected multipoint units (Figure 4) or either digital-to-digital loopbacks or analog-to-analog loopbacks at coder/decoders (Figure 5) to be operated under control of the master station without affecting downstream transmission.

In addition, such a network could be temporarily broken into regional networks (Figure 6), for example, for regional sporting events or regional commercials, all under control of the network broadcaster. Today, broadcasters use additional part-time transponders for this purpose. A regionalizable terrestrial network would diminish the need for additional part time channels.

It should be noted that a "keep-alive", framed-with-correct-parity,[13] DS3 signal must always be generated by the multipoint unit, on any port not carrying video (switched-off), towards any DS3 transmission facility to prevent false protection switching/ false alarms on certain embedded fiber and microwave systems. In addition, both multipoint units and decoders would have to respond properly to DS3 rate Alarm Indication Signals (AIS)[14] generated by fiber and radio transmission systems when an unrestorable failure has been detected; and coders and multipoint units could not be allowed to falsely emulate AIS signals.

Although a change from one coder as a video source to another coder as a source would result in a decoder having to digitally reframe, a broadcaster or affiliate would delay insertion into the local program until after the brief transient has passed and the picture has been picture-frame synchronized locally so the viewer would see no transients resulting from network reconfiguration.

A broadcaster could operate several such networks, for example, for back-up, for simultaneous news transmissions for later rebroadcasting or for promotional material.

Ideas for software control and networking will be further socialized with the ultimate users, to be sure that the flexibility, security and features meet their needs at the outset.

Further detail (Figure 7) shows how the DS3 rate video codecs, DS1 rate audio codecs (channel banks), DS3 rate multipoint units and analog extensions can be interconnected to create a network.

Synchronization - DS3/DS1

For a two-way multipoint network, the video coder design is simplified if the DS1 clock is derived from the coder's DS3 clock and the DS1 is loop-timed at the channel bank. This eliminates the need for complex asynchronous pulse stuffing/destuffing.

Control Channel

Two alternatives (perhaps both) are obvious. The first approach is to use one or more of the twenty four DS0's contained within the DS1 bit stream. Up to six (384 kb/s total) are available since 15 kHz stereo audio and an SAP consume only 18 of the available 24 DS0's. This approach has the advantage that it can be used in the loop plant, where a DS1 extension can readily be found, even if a DS3 rate loop is not yet available and an analog loop (e.g. A2AT) must be provided for video.

The second approach would be to use otherwise unused stuffing control bits (so-called C-bits) which occur at a rate of about 200 kb/s. This would have the advantage of less complex insertion/ extraction at the multipoint unit relative to the first approach but leaves a problem for situations where analog video extensions have to be used.

Costs

Using current prices for 9 bit DS3 rate coders ($50K one end 2-way) and an estimate of installed costs per DS3 channel mile on 565 Mb/s (12 DS3) fiber optic systems, the annual revenue requirements for a single, unprotected, 2-way, nationwide multipoint network using DS3's and multipoint units serving about 200 affiliates compare favorably with annual lease rates for single-video-per-transponder (one per direction) satellite systems. However, satellite costs increase with the number of regional networks (a transponder is needed for each regional program), whereas the terrestrial network described can be regionalized with no increase in costs. When the cost of protection (standby DS3's for fiber systems or standby transponders for satellite systems) is included, terrestrial systems which use 1 by N protection (N is large) are considerably more favorable than 1 by 1 protected transponders.

Even allowing for the higher cost of 45 MB/s coders relative to coders without bit compression operating at 139 Mb/s (DS4E rate), the higher per-mile cost of transmission more than offsets the lower cost of coders at distances exceeding about 70 miles. Furthermore, although fiber system interfaces at 139 Mb/s are feasible, they are not generally deployed in the United States and it is unlikely that digital microwave systems could operate at such a speed with current 64 QAM technology.

These costs are all moving targets, however. High quality DS3 rate coders have been recently announced for about $20K. Per-DS3-channel-mile costs of fiber systems may be nearly halved when gigabit systems become available carrying 24 or 36 DS3's per fiber pair. Satellite transponder lease rates are likely to change if the supply/demand equation becomes affected by too few transponders being chased by too much data, voice and video demand in a period of launching hiatus (United States and European launch vehicles are currently grounded and government launching of United States payloads is likely to be replaced by commercial launching) - a complete reversal of today's transponder glut situation[15].

On the other hand, larger capacity satellites (with both C-band and Ku-band transponders in a single satellite) will lead to lower per-transponder costs. In the future, analog technology could lead to two high quality video signals per transponder (used internationally today but not of acceptable quality domestically) rather than one.

In addition, transponders of 40 MHz per channel are said to be capable of 60 Mb/s digital transmission and high quality digital component coders operating at 30 Mb/s (hence two videos per transponder) have been demonstrated by the Japanese[16]. However, the 30 Mb/s system, obtains bit-rate reduction by leaving out color difference information on every other horizontal picture line, which may not be acceptable for domestic broadcast services.

Terrestrial networks will never be quite as flexible as satellite networks for multipoint distribution chiefly because, at inception, any receiver can aim at and tune to any transponder autonomously. However, with proper planning, a terrestrial network can achieve any desired degree of flexibility. Ultimately, it is expected that the two technologies, terrestrial digital transmission and either analog or digital satellite transmission, will be mutually supportive.

The Challenge

The challenge before us is to create a North American DS3 rate multipoint service capability with compatible coders, multipoint units, and an open architecture control network to permit a customized, perhaps user-provided controller.

Bellcore has been charged by its Regional Company owners to propose requirements for coders, multipoint units and controllers, and to make those proposed requirements available to all potential suppliers and finally to assist in a DS3 rate multipoint, multi-carrier, multi-supplier nationwide trial available for use by broadcasters. All are invited to participate.

This is an ambitious program which is expected to take about two years to complete. It includes: establishing proposed requirements, supplier development of prototypes, prototype testing for compliance, selection of nationwide sites for coders, and multipoint units, development of software for controllers, installation and testing, and a following soak period in actual video service. Participating suppliers must agree to bear their own costs, to meet established schedules and to terms of full disclosure of end-to-end compatibility details on a timely basis to permit other suppliers to proceed with development of compatible units. Once a standard is established, competition among suppliers would be in areas of size, price, power consumption, extra features, reliability, etc., but all products would still be compatible with each other.

Transition to Component Transmission

An international standard exists (CCITT Rec 601) for the digital component television studio with a companion standard (Rec 604) calling for component transmission of component-sourced video rather than composite transmission. There is also a SMPTE/EBU standard[15] for digital component video tape recording (Type D1). The bit-rate required to transmit an uncompressed digital studio output is 216 Mb/s for video, plus additional bits for housekeeping and audio. Transmission of such information, without compression, by a broadcaster to its affiliates, would consume the equivalent of about 6 DS3's and therefore would be about 6 times as costly per mile as single DS3 NTSC composite video transmission.

It has been conjectured that compression to 90 Mb/s or 2 DS3's would be acceptable to broadcasters for component transmission. If this is the case, then the DS3 networking described here-in could possibly be thickened to a second co-routed DS3 with the second DS3 slaved to the first via an upgraded multipoint unit; the same control software and control channel structure used on the first DS3 would suffice. The first DS3 could carry the luminance, DS1 for audio and embedded control channel; the second DS3 could carry the two color-difference signals for black and white survivability if only one DS3 is restorable in the event of a failure.

It is worth noting that a component coder is really three separate coders, one for each component (luminance and two color difference signals) and therefore likely to be significantly more costly than an equivalently complex composite NTSC coder.

Therefore a broadcaster must foresee some advantage to component transmission for distribution to affiliates to offset its significantly higher coding and transmission costs. There are obvious processing advantages for component transmission in the contribution phase; advantages in the distribution phase are less obvious in an environment of NTSC home receivers.

In addition, analog technology has advanced to the point where, through pre-filtering and/or post filtering, NTSC subjective artifacts can be reduced significantly. Therefore, an affiliate, receiving NTSC from a network broadcaster via composite transmission, could convert to component for processing if necessary, and back again to NTSC without appreciable additional NTSC artifacts.

Summary

DS3 rate (45 Mb/s) coders, multipoint units and controllers can be configured to provide broadcast quality distribution of NTSC television to affiliates on terrestrial, multipoint, digital fiber and microwave interoffice facilities in a manner rivaling today's satellite networks in cost, performance, reliability, flexibility and security.

With such a network, a broadcaster-to-affiliate network can be: reconfigured; regionalized for special events; monitored for error performance and picture quality at a Control Center without interrupting service; sectionalized for trouble without disturbing service; and if need be, changed to an alternate Control Center.

Bellcore, together with the seven Regional Companies will be working with the multi-supplier, multi-carrier and multi-broadcaster community in an effort to establish a standard algorithm for coders and an open architecture for control of multipoint units by broadcasters. A trial is planned by the end of 1988. All are invited to participate.

Acknowledgement

This paper draws upon discussions with coder technologists, in the U.S., Canada, UK, Italy and Japan and more than two years of rubbing elbows with network broadcasters at standards meetings and other meetings. It is also a result of work within Bellcore and with Bellcore's seven regional owner/clients. Individuals who have been most influential in developing the ideas are Al Goldberg and Sy Yusem - CBS, Ron Gnidziejko - NBC, Herb Riedel and Howard Meiseles - ABC, Ralph Mlaska - PBS, Peter Tu - AT&T, Bell Labs, Shaker Sabri-Bell Northern Research, Hideo Yamamoto- KDD, Yoshihiro Yamamura - NEC America, Yukihiko Iijima- NEC Japan, Silvio Cucchi - Telettra (Italy), David Crawford - British Telecom, Laurie Goody - Telecom Canada, Krish Prabhu - Rockwell, and Steffen Rasmussen - ABL.

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Switched Selection Of Remotes To Receive

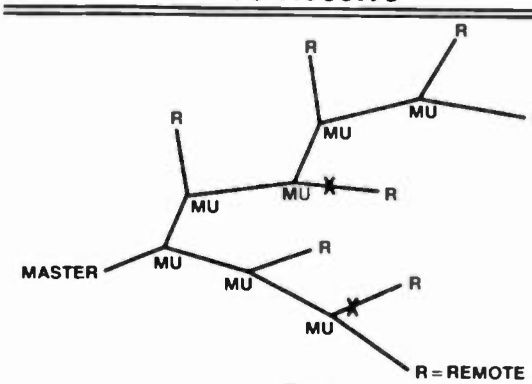

Fig 1

Switched Selection Of Remote To Send

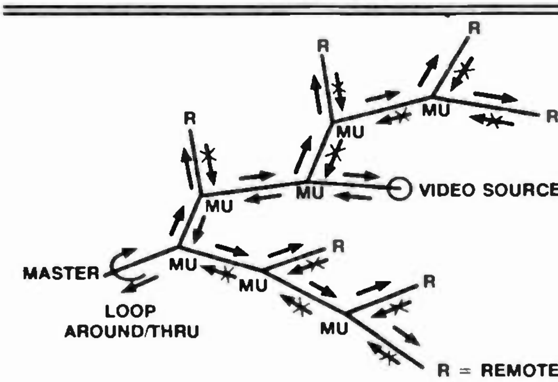

Fig 2

Change Of Control

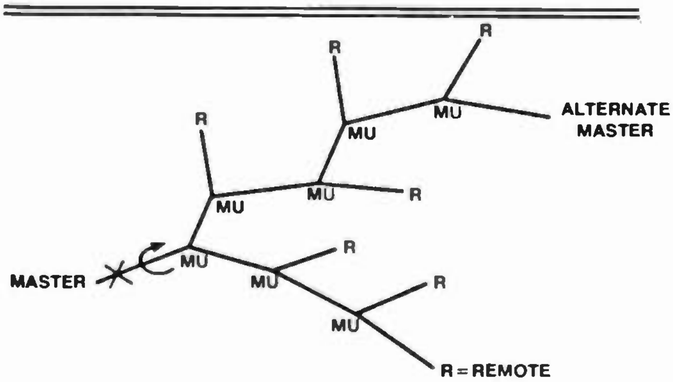

Fig 3

Addressable Loopback/Thru Test @ MU

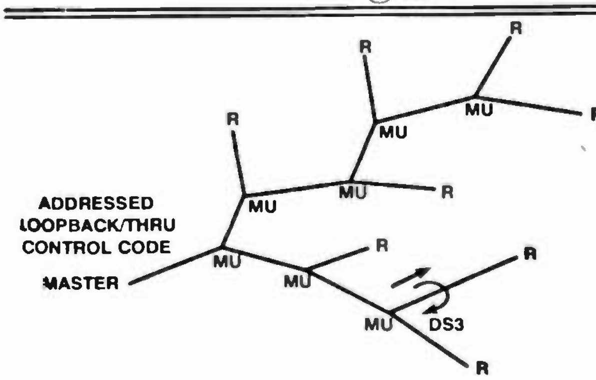

Fig 4

Addressable Loopback/Thru Test @ Remote

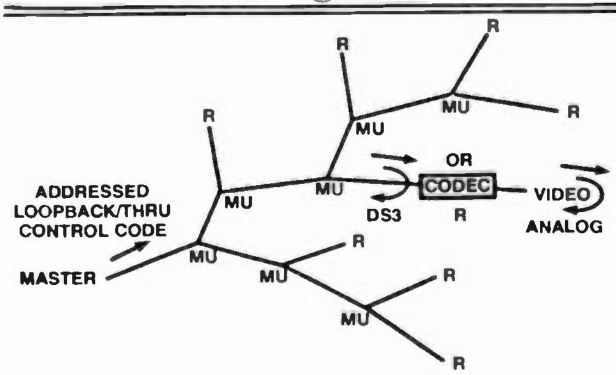

Fig 5

Splitting Into Regions

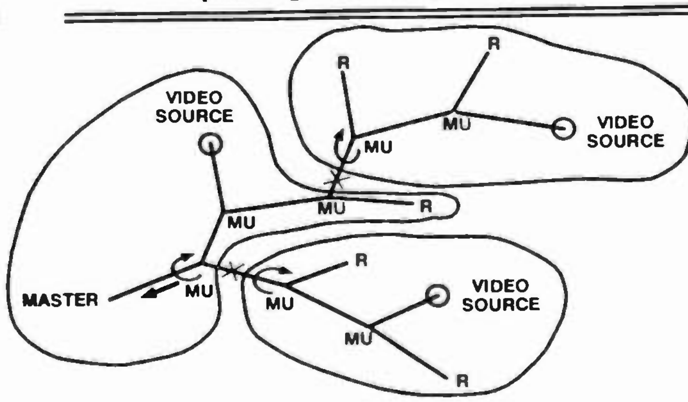

Fig 6

Detailed Network Configuration

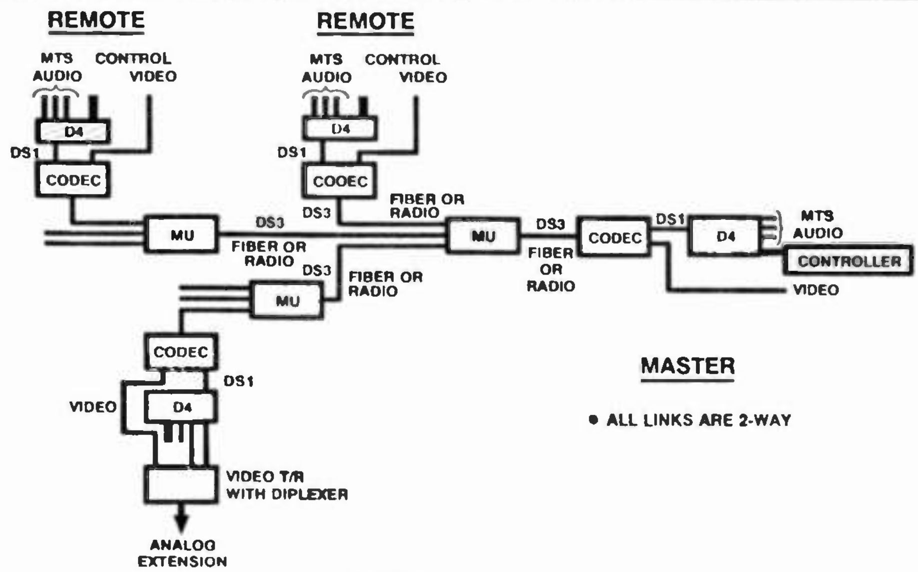

Fig 7

FIXED EARTH STATION DESIGN AND OPERATION FOR BROADCAST VIDEO—A SYSTEMS APPROACH

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This paper is written principally to discuss the key issues, which in the view of this author must be considered in the design, implementation, and operation of a satellite based broadcast video distribution system. And most particularly why a system approach to the task is needed. Since the time available for this paper will not permit discussion of the many actual system design issues involved, this paper is specifically intended to raise those issues which our experience has shown to be most important, particularly those operational issues.

The specific material presented in this paper results from experience gained by Group W Satellite Communications (GWSC) in dealing with technically sophisticated and operations-oriented clients, whose traffic primarily consists of high value network broadcast video programming, and who demand both state-of-the-art technical performance and reliability. Since GWSC is both a provider of such services, and a user of such communications services, I feel we are somewhat of a hybrid operation who understands both sides of the business.

Because Group W Satellite Communications is not as much of a household word as are the names of many of the companies you work for, let me digress for a moment to tell you who we are. GWSC is a business unit of Westinghouse Broadcasting Company, Inc., which in turn is a business unit of the Westinghouse Electric Company. GWSC has interests in a number of businesses which in general relate to cable television program services, and one business, the Operations and Engineering Group, which provides transmission services to third party clients. Specifically the Operations and Engineering Group provides four types of service. One service is the provision of playback, production, and satellite transmission services for Cable programmers (The Arts & Entertainment Network, Lifetime, The Discovery Channel, Request Television, and The Travel Channel are five Cable program services which are originated at our facility). The second type of service is satellite transmission service for Broadcast Networks and Independent Television stations (ABC, CBS, WNYW-TV, and numerous "occasional" transmission users are examples of such clients). We also plan to include satellite transmission capability for international voice, data, and video traffic in our service offerings. Additionally, we provide certain consulting engineering services for satellite based video transmission system design and operation.

The GWSC Stamford Complex is well suited to provide these transmission services, as we have a major playback, production, and transmission facility located in Stamford, as well as a satellite earth station complex which we believe to be the largest video earth station facility in the U.S. This facility currently employs some ten satellite earth station antennas which range in size from 11M to 5M (which service both C-band and Ku-band clients), we operate some eighteen 3.3 Kw High Power Amplifiers, and some 48 program receivers, all as a part of a completely integrated "broadcast quality" facility. The facility currently handles nearly 4000 hours of revenue video transmission service each week. Group W was a pioneer in the broadcast video satellite transmission business, as we installed five full C-band uplink/downlink systems (one at each of the five Group W television stations) beginning in 1980.

In the first part of this paper, we will deal specifically with what steps we feel are necessary to consider, to plan, design, construct, operate, and maintain a transmission system which provides the level of service required by a broadcast/cable network client. The system which this paper is based upon is a combined terrestrial microwave/fiber optic system and satellite earth station system which carries traffic for multiple video users between several midtown Manhattan locations and our satellite earth station complex. This transmission system, which was designed and is operated in large measure to service broadcast and cable network clients, makes a good example for this paper, as we are dealing with providing broadcast quality video service on an end-to-end basis.

The system design, as I believe all broadcast system designs which contain an earth station operation should, centers around four principle parameters: reliability, performance, cost effectiveness, and flexibility. In the case of reliability and system performance, the EIA Standard RS-250B specifications provide the design standard. This specification provides guidance for both system technical performance and for system reliability. However, several of the RS-250B parameters require tightening or augmentation to accommodate the need for the system to pass encrypted material, and for the requirements of multi-channel sound.

System reliability should be the major design parameter in the broadcast business, and particularly in the Broadcast Network distribution business. Continuity of service is the name of the game. Broadcast air time is irrecoverable. When you lose time you lose both revenue and audience. The standard of continuity of service is 99.99%, which equates to approximately 53 minutes of outage per year of full-time service. This standard requires a significant management effort to meet, particularly when typical traffic is not continuous and multiple switches of facilities are required.

The next point is system technical performance. When the technical quality of the programming that your clients audience receives is principally controlled by the distribution system you run, and it's a very competitive business, you should expect your distribution system to be essentially transparent.

Cost effectiveness is the next point. On top of the rather exacting reliability and performance requirements, the system has to be cost effective. In practice this means that the always fine line between the quantity of standby equipment and the number of operations and maintenance personnel needed to maintain the needed level of performance, and the cost of providing the service is still there. What this really means is that the initial system design must be very carefully analyzed to ensure that the overall objective can be met and still allow for a profit. We relied heavily on our previous experience in satellite uplink operations and rigorous Mean Time Between Failure (MTBF) calculations for each piece of equipment to determine the criteria used for the final system design.

The last major point is operational flexibility. While the broadcast business has been around for some time, the business continues to evolve. What was yesterdays standard operating procedure may not be applicable tomorrow. It is, therefore, very important to factor into the system design the ability to accommodate both operational and equipment changes; specifically, planning should allow system changes to be made in such a way as to minimize the impact on ongoing operations. Consequentially, we designed the system to provide maximum flexibility.

With these four key technical requirements as background, we can now talk about how, in our view, such a system should be designed and implemented.

In a transmission system operation, which is essentially the service we provide to the broadcast and cable networks, system operational reliability is really the key design parameter. The three other factors; performance, cost effectiveness, and flexibility really follow reliability when it comes to items in the system design where the initial design decisions has such long term impact.

Reliability, from a design criteria, really consists of three major elements. These elements are: propagation, equipment, and personnel. From our experience we based our design on the following allocation of the approximately 53 minutes per year of outage contained in the initial assumption of 99.99%. We first allocated 50%, or 26 minutes, to the technical system, and the second 50% to personnel errors. Remembering that this system, for these clients, is an operations intensive system. The technical system budget is then further broken down to provide 25%, or 13 minutes, of the total to equipment failure and the second 25% to propagation. This assumption has proven to be quite accurate, as our reliability experience over the last twelve months closely matches the predictions. In actual practice, we have found that equipment problems occur more frequently but tend to be of short duration, while personnel errors are less frequent; but typically result in relatively longer service interruptions.

In order to meet the design objective of approximately 13 minutes of annual outage due to equipment failure, we did a full reliability study on all the hardware utilized in the system. This study dealt specifically with system Mean Time Between Failure (MTBFs), and developed both the quantity of spare parts we keep in inventory and the maximum permitted equipment repair turn-around time we contractually obligated our suppliers to provide. The study also indicated we would have to have sufficient maintenance staff and parts available, to accomplish an annual 6 hour Mean Time to Repair (MTTR) performance for the system.

In treating the personnel factor, we spend a considerable amount of time training our console technicians (our operations people), and we also very carefully select these individuals. The responsibility our console technicians have is perhaps unique in this business, as these technicians have to be competent with traffic, operations, technical parameters, and know enough of how the system works to be able to make considered judgments on how best to work around problems and maximize the traffic capability in the system. Training and retraining is the key element here. In complex plants people must have ongoing training to keep their efficiency at the high levels needed for reliable performance.

System performance requirements are quite rigorously defined in EIA standard RS-250B. However, when the 250B standard was developed, certain matters such as encryption, multi-channel audio, and 18db of audio headroom were not treated. All of these matters are quite important to the present day network transmission operations, so suitable specifications for these parameters were added to the system design as well. To meet these parameters it is necessary to treat the complete system, both microwave/fiber and earth station, as an end-to-end system; and to develop an impairment budget to specify the necessary performance for each segment of the system.

Considerable care had to be exercised in the system design and implementation, particularly with the video signal-to-noise (S/N) parameter. We elected to specify a per hop objective for the terrestrial system of approximately 73db, to maintain a end-to-end video S/N of 68db. The additional requirements of being able to handle both NTSC formatted material and encrypted formats adds another design consideration, as does the ability of the system to handle stereo programming with the inherent interchannel phase and gain criteria. It is desired that the transmission system be essentially transparent for both audio and video traffic.

The next point in the design parameters is "cost effectiveness". To keep costs within budget we carefully analyzed such items as the optimization of protection equipment, we allowed space for efficient expansion, and we provided the needed physical space, test points, and monitoring to allow for efficient maintenance. Being cost effective for the hardware required for the initial system traffic requirements during the initial design phase, while frequently requiring certain trade-off's be made, is relatively easier to handle than that required when looking down the road. Trying to balance initial system implementation costs against the costs required for items related to future expansion can be difficult. The facilities, space, power, etc., required for a possible future need can be difficult to justify; but if not initially provided for, future expansions can be both very expensive and also be disruptive of ongoing services. Our rule of thumb is to provide for, at a minimum, a 25% expansion capability to accommodate future traffic requirements.

A side point here, that perhaps bears mentioning, is the importance of dealing with the actual physical construction of the system as a part of the initial design considerations. The principle points here are low maintenance, security, and expansion capability. The routine aspects of building design, HVAC, and antenna mounts should to be considered as well, of course. It doesn't help the project if equipment is planned to go into an eight-foot rack, then to find out during the installation that you have to deal with a seven-foot high elevator. If you don't take anything for granted, you won't have any unpleasant surprises.

The last point in the design is dealing with operational flexibility. This flexibility goes all the way from hours of operation, to transmission deviation standards, numbers and characteristics of audio channels, and transponder and satellite assignments to name but a few. Operational flexibility, like expansion capability, can be difficult to deal with during the initial design phase. System design, by nature, is much easier to deal with when all the parameters are stated up-front. However, in all broadcast/cable systems allowance has to be made for change. Accordingly, we recommend that patch panels be generously used; that, where possible, control systems be software driven and that hard-wired controls be kept to a minimum; and full matrix switching systems be used for both baseband and R.F. systems to the extent that the increased initial cost can be justified. In our experience, built-in flexibility is always needed and is fully justified.

Now that system designs parameters of reliability, performance, cost effectiveness, and flexibility have been reviewed, it's appropriate to review what, in our experience, must be considered in developing the system operational considerations. The one major area where a fixed earth station operation is unique, is with actual operations (i.e., how do you actually operate an earth station complex to handle the traffic required with major broadcast/cable network clients), clients who expect service which is responsive, accurate, and reliable, with very high technical quality? At GWSC we work to meet these operational requirements by dealing with these points: Top quality people, functional operating positions, standard operating procedures, well designed and maintained hardware, and last but certainly not least - training.

Top quality people are generally people who have had previous experience in the broadcast business, either in a technical or traffic capacity. They are bright, concerned with the business, happy to receive training, and willing to accept responsibility. We find that it typically takes one-to-two years to fully train our earth station console technicians, and they are consequentially among the higher paid individuals in the technical operation. In our case, we also have a traffic technician on duty to handle and schedule service orders.

On the point of functional operating positions, our operating positions are designed to be worked by more than one technician if traffic activity requires an extra set of hands. The positions are also designed for maximum user efficiency, so all needed operations can be accomplished without the operator moving from the basic operations console. Two basic positions are established, operations and traffic. The operations position is where the earth station technician works, and that person has the overall responsibility to run the earth station complex. Specifically, this technician is responsible for all uplink operations, which include antenna movements, frequency changes and power level adjustments, dealing with the satellite carrier for cross-pole isolation and saturation power level checks. The same general matters are handled for downlinks.

The traffic person has the responsibility to take the orders, schedule the facilities, and deal on a real time basis with the clients. The traffic person also has the responsibility to check the "quality" of the traffic material from a technical standpoint and "sell" it to the client. Additionally, the traffic person has the responsibility to deal with the business aspects of the business; aspects such as client identification and billing information, any credit problems, and, perhaps most importantly, to work to optimize the amount of traffic that is being handled through the system.

In all cases, the operating positions have been designed to minimize the number of people required to provide the client with the required level of service. When the business volume gets heavy at one or the other of the positions, the second person has been sufficiently cross-trained to allow that person to lend a hand.

Standard Operating Procedures (S.O.P.'s for short) are firmly and formally established for all operational issues, both normal and abnormal. We always want the technicians to fully understand how a specific function is to be performed. Establishing uplinks are perhaps the most formal of all of our procedures, either due to the requirement of the carriers or GWSC. Prior to and throughout the duration of an uplink, we tune a receiver to the relevant transponder and satellite, we also always watch the carrier come up on a spectrum analyzer, cross-polarization and saturation power levels are checked rigorously. Since we can have as many as fourteen uplinks occurring simultaneously through five antennas, we obviously have to keep on top of what is happening in the plant. The appropriate operating logs, discrepancy reports, and defective equipment reports are also kept as a matter of course.

On the hardware side, the system was designed specifically with the operator in mind. The computer control systems have the necessary "smarts" to inhibit an operator making certain classes of errors, and all the needed protection is automatically accomplished without operator intervention. We tried very hard not to build any "traps" into the operations systems. With the heavy volume of traffic we handle, the operations person shouldn't have to be concerned with system problems.

Maintenance is also handled in a rigorous and routine manner, with all maintenance functions to be worked around traffic requirements. Again, we do not want equipment to be out of service, or under test, in any way that would cause an operations problem. The operations, i.e., providing the scheduled service to our client is the top priority within the facility. This coordination between operations and maintenance can sometimes be difficult, as ongoing testing is needed to keep the plant operating so its full design specifications are met.

The last item, which is in many ways perhaps the most important, is the measurement of "quality" performance, and the using of such measurements to improve the overall operation. We rigorously measure performance. This measurement is principally done from the perspective of the client, i.e., we specifically measure the frequency and duration of any impairment to a clients service caused by GWSC. And we define impairment very broadly, it does not make any difference to us whether an impairment is caused by a traffic error, an operations error, or a piece of equipment failing. In 1986 the GWSC transmission operation had an overall reliability percentage of 99.995%. That number was developed from an average of 13,400 hours per month of revenue transmission service. We are very proud of that result, but there is always room for improvement, and we try to do just a little better each month.

To do just a little better each month, we provide training, establish goals, and give employees real-time feedback on their performance. You can't provide quality and reliable service, if the employees do not know how to handle the plant. We provide a rigorous initial training program and ongoing reviews with all personnel to ensure that they know how to do their jobs. We are also very fortunate to have good people, who do a very good job. I believe the bottom line to a quality fixed earth station operation is with the people. You can install the very best most reliable equipment available, but if your people don't understand the operation, particularly during periods of abnormal operation, you will not be able to provide a top quality level of service.

To repeat these five points: To run a quality operation, you need good people, the operating positions have to be designed to facilitate the level of operations contemplated, standard operating procedures need to be in place, you need well designed and maintained hardware, and you need an ongoing formal "quality" program to ensure that your service needs are being met.

So to conclude, in our view, the key points that should be specifically considered in the initial design of a transmission system are reliability, performance, cost effectiveness, and flexibility. Then to operate the facility you need good people, effective operating positions, standard operating procedures, and perhaps most importantly a formal "quality" program to insure that the requisite service levels are being maintained.

All of the above operational points are really simply common sense; but, the system design points were developed from both standard good engineering practice and our own experience. With a large system, you have to work at being good all the time. As it seems things do not get better by themselves, they only seem to get worse. So my final comment is that if you want to maintain a top quality and reliable operation, keep after it. It's always better to handle a problem today, don't wait until tomorrow. Again, this is common sense, but it's common sense that sometimes seems to be ignored.

COMPUTER TECHNIQUES HELP SOLVE SATELLITE EARTH STATION SITE DESIGN PROBLEMS

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INTRODUCTION

Many earth station sites involve multiple antennas. At such sites, it is desirable to place the antennas as close together as possible in order to take advantage of economies of land use and minimize waveguide loss. Shielding from terrestrial interference is required at many C-Band sites. This can involve placing earth stations close to existing buildings or constructing shield walls in close proximity to earth stations. Figure 1 shows an example where shield walls were built on a limited size rooftop site. Figure 2 shows a site with concrete shield walls. Many sites have existing obstructions such as buildings or trees nearby. In situations such as these, the designer must plan the site with great care.

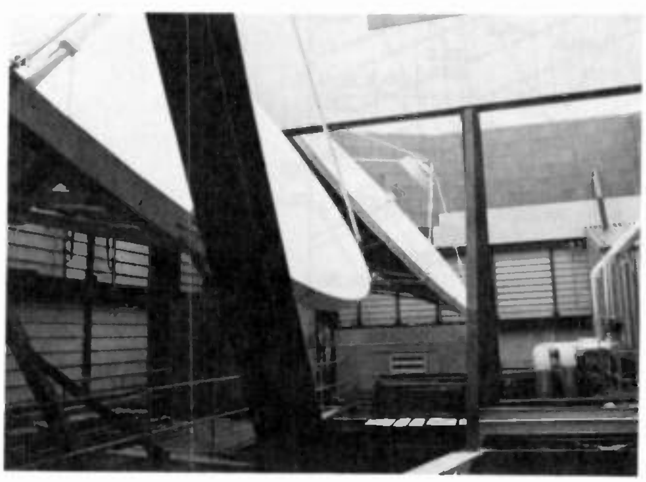


FIGURE 1

ABC NEW YORK CITY UPLINK

Two 9.1 meter C-Band uplink dishes have been placed on the roof of ABC's new technical center in Manhatten. Frequency coordination with terrestrial users required that walls 31 to 36 feet high be placed around the antennas to form a box 60 x 100 feet. Microwave absorber was applied to the inside of the walls, which are as close as six feet to the dishes.

FIGURE 2

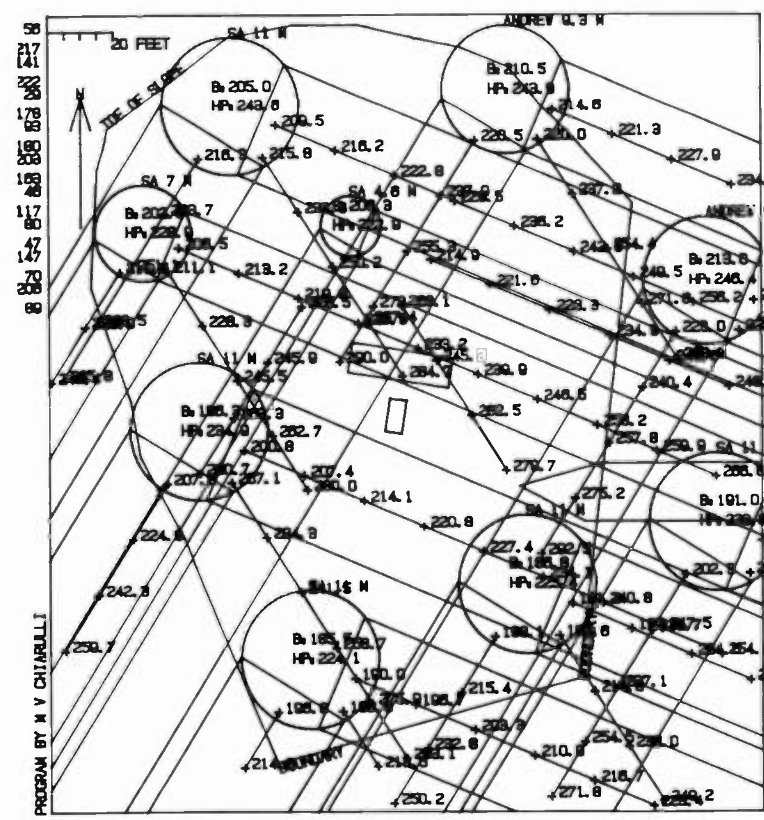
WBNS-TV EARTH STATION

Four C-Band satellite antennas at WBNS-TV's satellite earth station located at their studios in downtown Columbus, Ohio, are protected from radio frequency interference by a 31-foot high, 740-foot long modular concrete wall and earth berm combination designed by Fanwall Corporation. The wall, which consists of 20-foot by 9-foot precast concrete panels joined together with aircraft-type steel cables, is entirely free-standing.

Careful planning of sites requires a detailed knowledge and calculation of the geometry of earth station antennas. This includes the determination of the azimuth pivot point, the elevation pivot point movement, actual reflector location, look angles, and maintenance positions. A detailed knowledge and calculation of the geometry of the beam to the satellite is also required. This includes the beam location and height, and for very close work, the shape. Detailed information on shield and obstruction shapes and locations is also needed, as is site topography.

Even using certain applicable simplifying assumptions, cases involving placing two or three antennas in a limited area are very tedious to solve. One solution to this problem is to use a computer to model the detailed geometries discussed above. Then, if the results of a first guess at locations is plotted, it oecomes easy to modify the plan through successive iterations. This way the designer can consider many arrangements in order to find the best solution given the site limitations.

Figure 3 shows a site plan for the placement of nine antennas on one site at El Cerrito, California. In this case, approximately ten proposed solutions were presented one after another over a four month period. Each solution involved approximately twenty-five iterations to produce. The successive solutions were required mainly by changes in assumptions about the expected size of future antennas. At one point, a major reworking was required when soil borings indicated that some areas in the site were unsuitable for development. With all the antennas, the site topography, and the site obstructions already modeled in the program, the almost totally new arrangement was produced in one day.



EL CERRITO EARTH STATION SITE PLANNING

FIGURE 3

EL CERRITO EARTH STATION SITE

Plan developed to place five 11 meter, two 9.3 meter, one 7 meter, and one 4.6 meter antenna on naturally shielded site of limited size in El Cerrito, Calfiornia. B denotes the base elevation of the antenna based on site topograph modeled in the computer. The box-beam and cylindrical assumptions have been used.

The three sections that follow will describe the antenna geometry, beam geometry, and shield and obstruction modeling. Simplifying assumptions and examples will be presented within the sections. Formulas for calculations will be presented in an appendix. The BASIC language programs used are not included because they are not properly documented. Furthermore, the detailed calculations could be ignored if one were to use a three dimensional computer aided design and drafting program. The recent availability of such programs for personal computers suggests easier ways to accomplish the actual plotting. The geometric modeling remains the same.

ANTENNA GEOMETRY

Elevation-Over-Azimuth Mounts

The typical earth station antenna has an elevation-over-azimuth mount. As can be seen in figure 4, the antenna reflector is attached to a hub which pivots on pins or hinges to make elevation adjustments. These hinges form an elevation pivot on a horizontal line. The frame that supports the elevation pivot rotates on a vertical shaft of some sort to provide azimuth adjustment. The center of this shaft is the azimuth pivot. From figures 4 and 5, it is clear that the elevation pivot point will rotate around the azimuth pivot. Thus to find the position of a point on the dish for a given azimuth, AZ, and elevation, EL, it is necessary to first find the position of the elevation pivot.

Elevation Pivot Location

The position of the elevation pivot is a function of azimuth and of the distance from the azimuth pivot to the elevation pivot, EPD (which is fixed for a particular dish). Thus, from figure 5, the distance from the azimuth pivot to the elevation pivot in the x-direction, EPD_x, and in the y-direction, EPD_y, are a function of azimuth, while EPD_z is the fixed height of the elevation pivot.

Center of Dish

Taking point M as the center of the circle defined by the edge of the reflector, we see from figure 6 that the distance from the elevation pivot to M in the direction of the azimuth, MD_{az}, is a function of elevation. Taking this dimension back into figure 5, we can find MD_x and MD_y. MD_z can be found directly in figure 6. The location of point M for a given azimuth and elevation is:

$$(EPD_x + MD_x, EPD_y + MD_y, EPD_z + MD_z).$$

Edge of Dish

Now, let E be a point on the edge of the reflector, and let q be the rotation of the reflector needed to bring E to the lower lip of the dish.

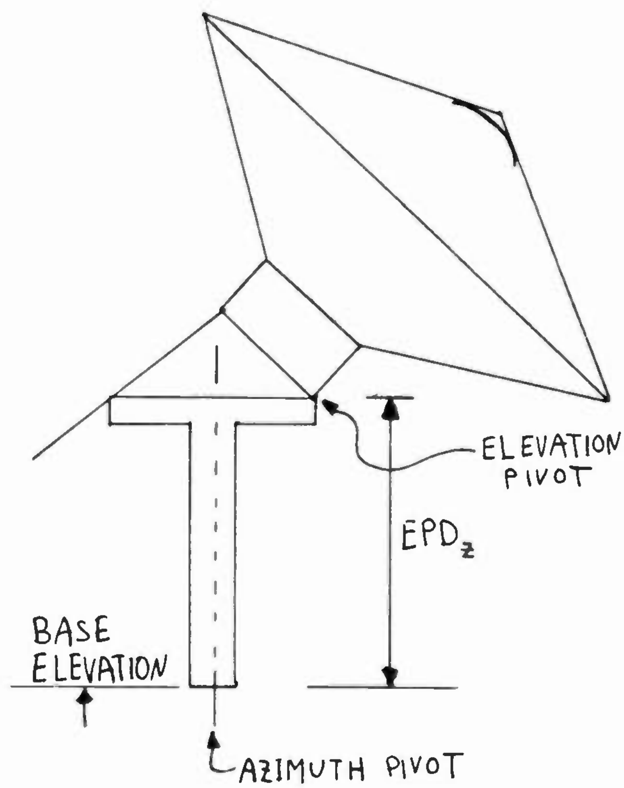


FIGURE 4

ELEVATION - OVER - AZIMUTH MOUNT SIDE VIEW

The relationship between the elevation and azimuth pivots is shown.

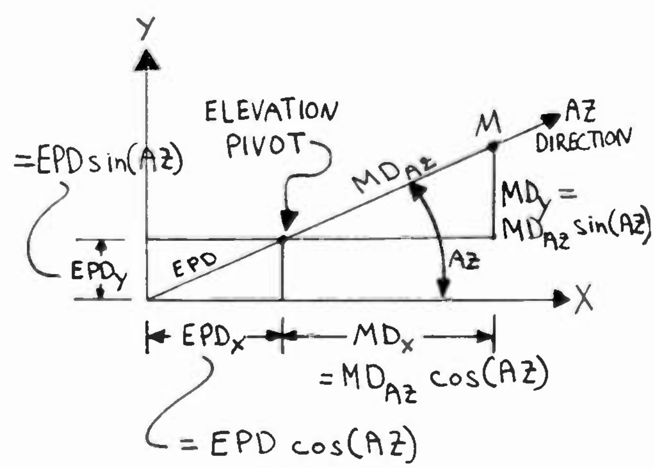


FIGURE 5

ELEVATION - OVER - AZIMUTH MOUNT PLAN VIEW

The relationship between the elevation and azimuth pivots is shown in plan view when the azimuth does not correspond to the x-axis. The location of point M is also shown.

Figure 7a shows points M and E in the plane of the dish, that is, looking directly into the dish from the direction in which it is pointing. R is the radius of the reflector, and k and n are the Cartesian components formed from the polar coordinates of R and q. Figure 7b is a side view of the dish showing how k can be split into a k_z and a k_{az} component. Using a plan view of the dish (Figure 7c) we see that k_{az} is the plan projection of k and that n is its own plan projection as it is already parallel to the ground. The components k_x, k_y, n_x, and n_y from figure 7c and k_z from figure 7b allow us to locate the edge of the dish. Watching the signs carefully, the location of a point on the edge of the dish is:

$(EPD_x + MD_x + k_x + n_x,\ EPD_y + MD_y + k_y - n_y,\ EPD_z + MD_z - k_z)$.

Figure 8 shows a computer generated plot that displays a plan view of the edge of the upper half of a 3.7 meter antenna. The antenna is shown at the azimuth and elevation for the look angle to a particular satellite. The look angle calculation was imbedded as a subroutine in the plotting program for ease of use. The numbers on the plot are the elevations of the points over the elevation of the base of antenna. This defines the exact location of the reflector and can be used to determine if the beams for other antennas (to be modeled in the next section) can pass over the reflector.

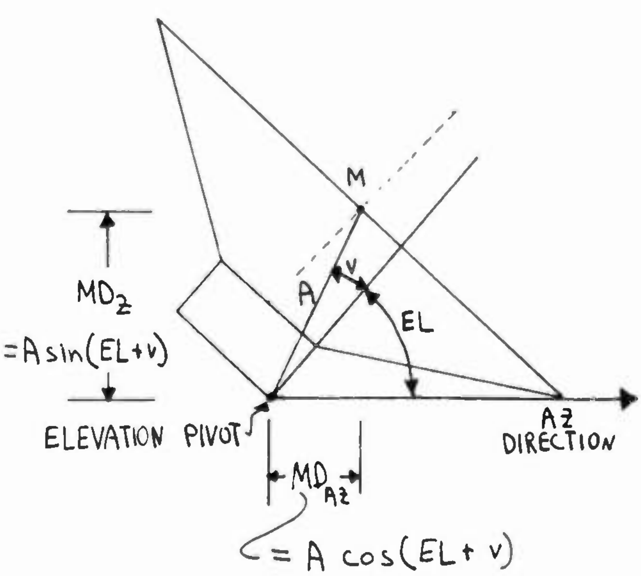


FIGURE 6

REFLECTOR DETAIL, SIDE VIEW

The location of point M is shown for a typical reflector.

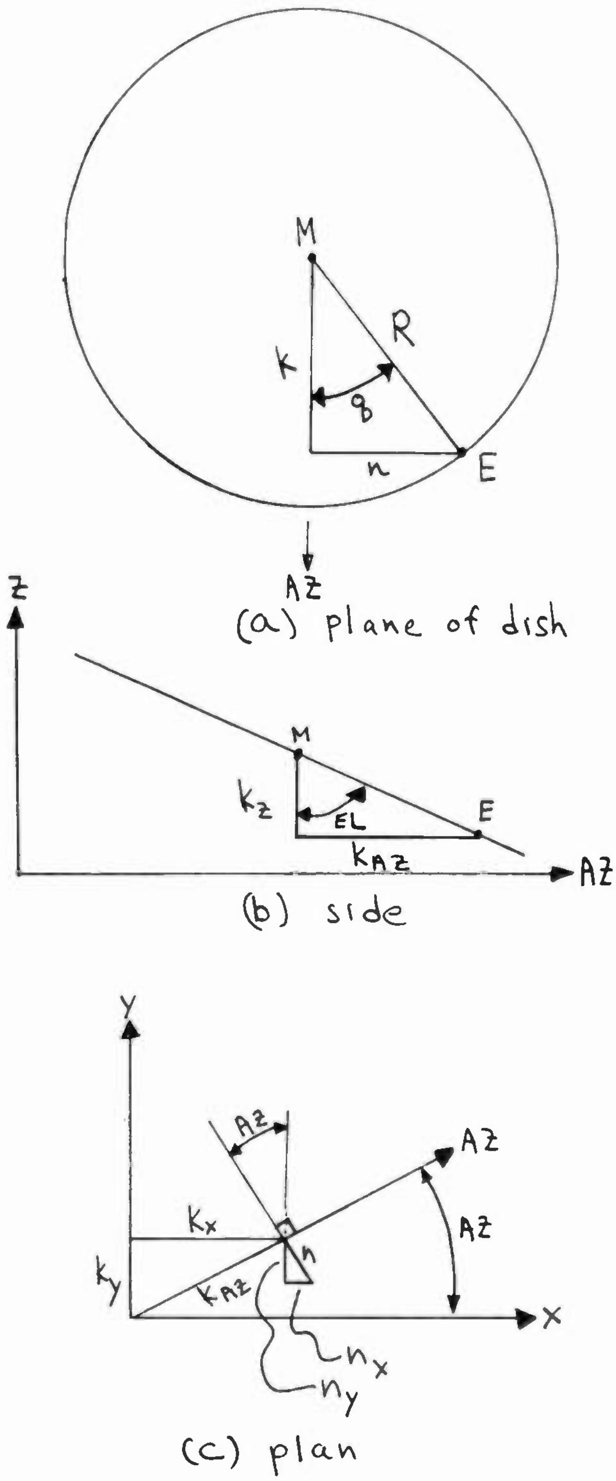


FIGURE 7

LOCATION OF REFLECTOR EDGE

The location of the reflector edge with respect to point M is shown. Figure 7a shows the view in the plane of the dish, that is from the satellite. Figure 7b shows side view, while figure 7c is a plan view.

Maintenance Position

A word or two of warning is appropriate concerning the maintenance position of antennas. In addition to planning for the particular look angles to be used it is important to consider other positions that the dish may be placed in for maintenance purposes. In a very tight site, limitations on maintenance could be accepted. In an extreme case, working on one dish would require the shut down of the dish behind it. In most sites sufficient leeway exists for the designer to allow complete freedom of movement to each dish without having an impact on other dishes. In fact, I would always start with this assumption and only discard it if a solution could not be found.

Cylinder Assumption

One simplifying assumption that can be used in modeling antennas allows for complete freedom of movement of each dish. This is called the cylinder assumption because the dish is assumed to occupy a cylindrical volume of space. The center of a circle forming the base of the cylinder is the azimuth pivot. The radius of this circle is not the radius of the reflector; it is a distance equal to the furthest possible extension of the dish. This is usually the lower lip of the reflector at some particular elevation. On some smaller dishes, the feed horn at zero degrees elevation can be the determining distance. The height of the cylinder is the highest possible extension of the dish. If this cylinder is not intersected by a beam from other antennas, then it is clear that the antenna inside the cylinder can be set at any azimuth from 0 to 360 and from any elevation from 0 to 90. This assumption was used in the El Cerrito site analysis shown in figure 3. The circles in that figure are plan views of the cylinders, while the maximum heights listed for each dish are the top of the cylinder above sea level. Due to the site topography at this site, the base elevation for each antenna differs depending on its location; thus the maximum heights differ accordingly.

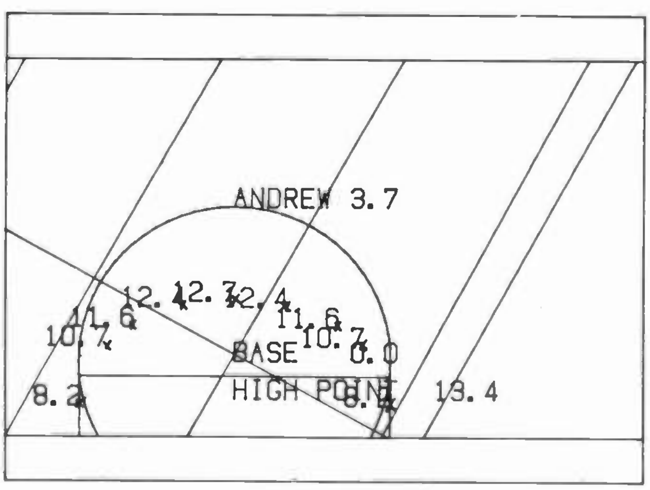


FIGURE 8

UPPER EDGE OF DISH
ABC TV 18/19 SITE

Computer generated plot of the upper edge of 3.7 meter dish. The points indicated denote the location in this plan view, while the numbers give the elevation over the antenna base.

BEAM GEOMETRY

Dependence on Antenna Geometry

The beam geometry is very simple given the antenna geometry. Using the edge of the dish locations developed in the antenna geometry section, the beam boundaries are simply lines extending out from the antenna edge at the proper azimuth and elevation for the satellite in question. Figure 9 shows the components for these lines, based on a repeated arbitrary plan view increment, I, from a point E on the edge of the dish. Plotting the beam location is simple on a computer where each increment can be plotted by a subroutine.

Figure 10 shows a plan view of the bottom of beams plotted on a computer. The beam elevation values show the height of the beam at the point indicated, while the beam edge lines show the width of the beam, without indicating the height.

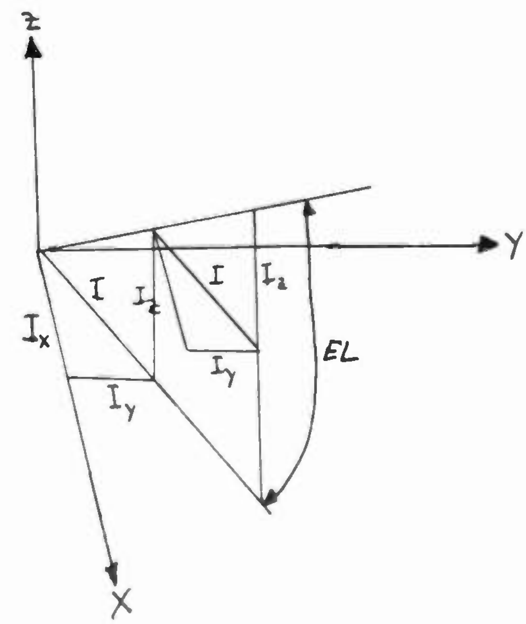


FIGURE 9

BEAM TO THE SATELLITE

The geometry of lines leading from the reflector edge to the satellite is shown for an arbitrary increment, I.

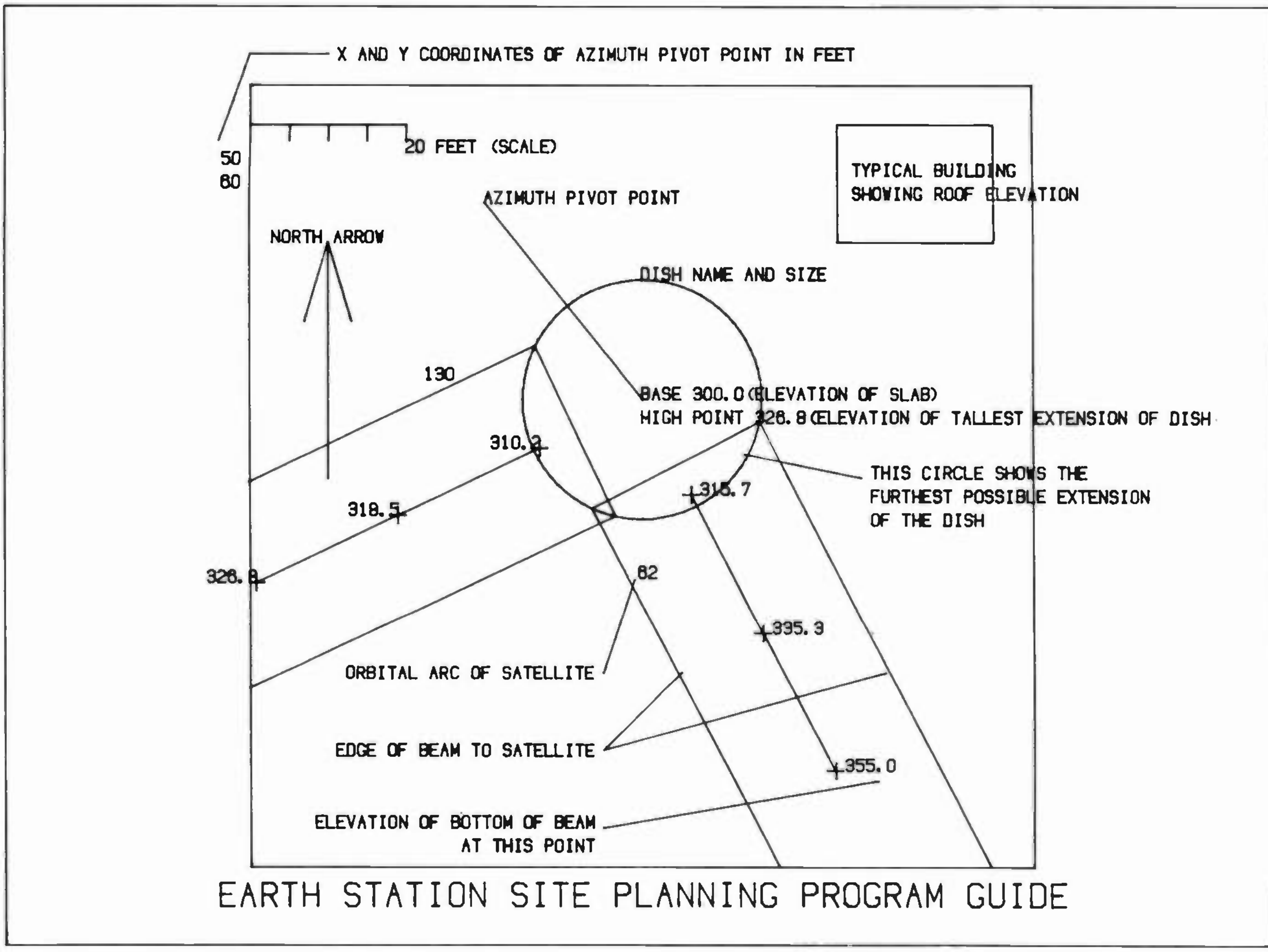


FIGURE 10

SITE PLANNING PROGRAM PLOT GUIDE

A guide to the site plotting computer program developed by the author.

Beam Shape

The cylindrical shape of the beam can be indicated by plotting the beam edge corresponding to several points on the edge of the dish. Figure 11 shows the bottom half of the beam edge for a 5.6 m antenna at a particular look angle. As in figure 10, the lines show the location of the beam while the values show the elevation at the point in question. Also shown in figure 11 is the dish edge of the 3.7 m dish shown in figure 8. There would be blockage of the 5.6 m dish if this site were built as shown because the lower beam edge of the 5.6 m dish passes below the upper dish edge of the 3.7 m dish. This particular case was an extremely limited roof top site where it was advantageous to use an existing steel dunnage. Thus, instead of moving the dishes further apart, the 3.7 m dish was moved down by shortening its mount by two feet. This imposed a penalty by limiting the orbital arc obtainable by the 3.7 m dish. As this penalty was deemed acceptable, the site was built accordingly, as shown in figure 12.

"Box-Beam" Assumption

Even with a computer, it would be tedious to plot a range of orbital arcs for two antennas as was shown for two particular orbital arcs in figure 11. Checking seven orbital arcs for each dish would require 49 plots like figure 11. A simplifying assumption that can be used to great advantage is to assume that the beam is box shaped. That is, for the full width of the beam, assume that the beam is at the elevation of the line from the bottom edge of the dish.

Figure 13 shows a "box-beam" passing over a cylinder-assumption dish. Using these two assumptions will allow work to proceed quickly as several box-beams can be plotted on one sheet without causing confusion. When necessary, full beam shape and reflector edge shape analysis can be made for the few cases that don't quite pass the box-beam/cylinder assumption.

SHIELDS AND OBSTRUCTIONS

Distant Obstructions

Shield walls and obstructions planned or occurring at some distance from the antennas can be modeled by adding lines to the site plots and noting the elevation. Single trees can be represented by cylinders, provided room is allowed for future growth. A line of trees is best represented by a vertical wall. Figure 3 shows a plan view of a small equipment building that was planned for El Cerrito, along with its roof height.

Close-In Shield Walls

When walls are built to provide shielding of earth stations from terrestrial interference, they are often built extremely close to the antenna. In cases like those shown in figure 14, a detailed determination of where the beam passes over the wall becomes critical. One way to make this determination is to calculate the location where the beams from reflector edge points pass over, or through, proposed wall locations.

With point E already determined from the antenna modeling we find in figure 15 the distance to the wall defined by angle s and distance T. This distance is I_{az}. From I_{az}, we find the elevation of the beam above edge point E using figure 9 to be:

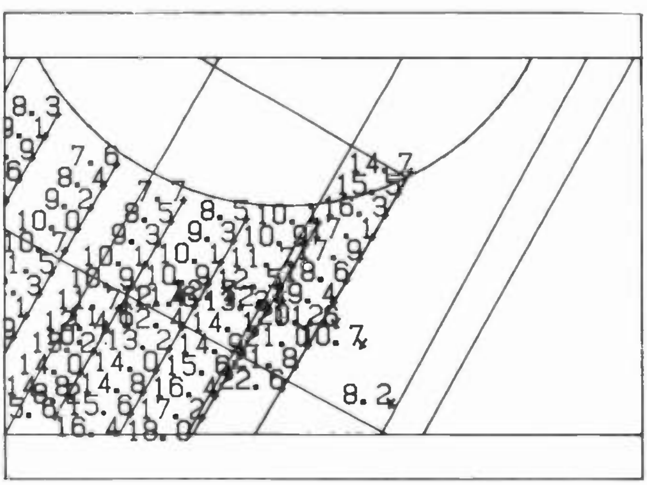


FIGURE 11

LOWER EDGE OF BEAM, ABC TV 18/19 SITE

Parellel lines show the path to a satellite from several points on the lower edge of a 5.6 meter dish. The points indicated denote the location of the elevation numbers given. The upper edge of the 3.7 meter dish shown in figure 8 is also shown.

$$I_z = I_{az} \tan(EL)$$

or

$$EPD_z + MD_z - k_z + I_z.$$

The distance from point O, shown on figure 15 to this intersection point is $T_w + n_w$. It is now possible to repeat the calculation for different points E and then for different look angles to calculate the clearance of the beams over proposed close-in shield walls. Figure 16a shows a plot of a shield wall with a top angled to track the look angle to the satellite arc. figure 16b shows the wall as built.

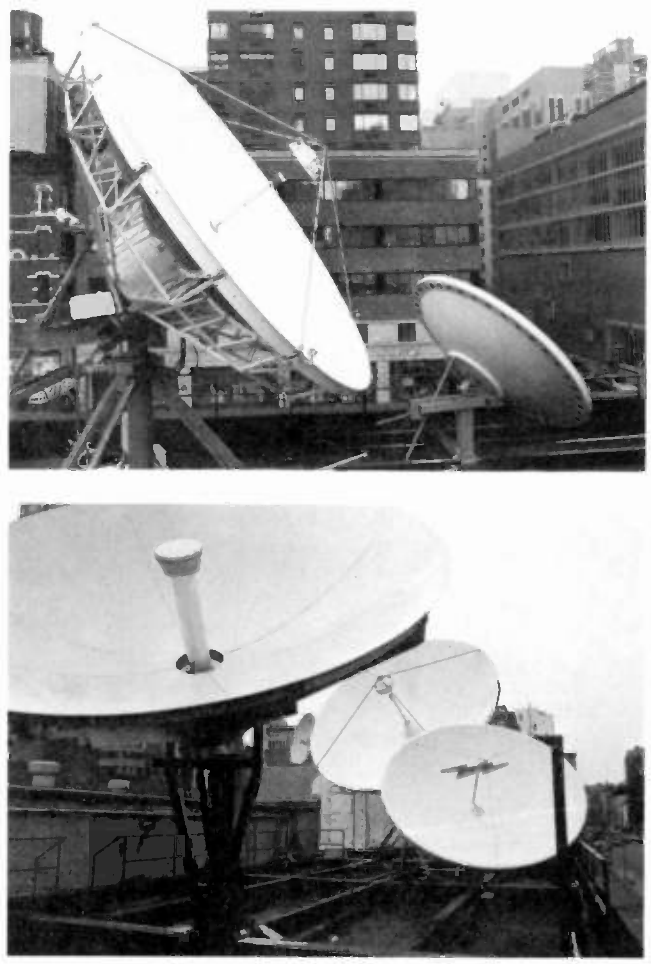


FIGURE 12

ABC TV 18/19 KU-BAND SITE

The ABC TV 18/19 site as built. The side view shows the 5.6 meter and 3.7 meter antennas. The front view shows, from front to back, a 4.6 meter, a 3.7 meter, and a 5.6 meter antenna. Also visible on a higher roof is a 4.5 meter C-Band dish used to receive the ABC radio network for monitoring purposes.

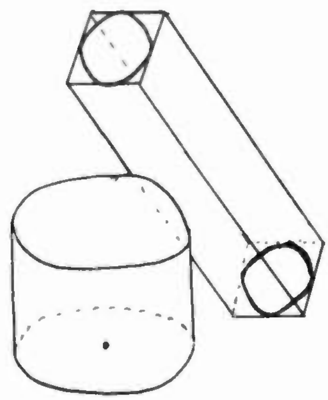


FIGURE 13

BOX-BEAM AND CYLINDER ASSUMPTIONS

A box-beam is shown with its inscribed beam indicated. A cylinder containing a dish is shown below the box-beam. These assumptions greatly reduce calculations and plotting. Details of the beam and reflector shapes are lost.

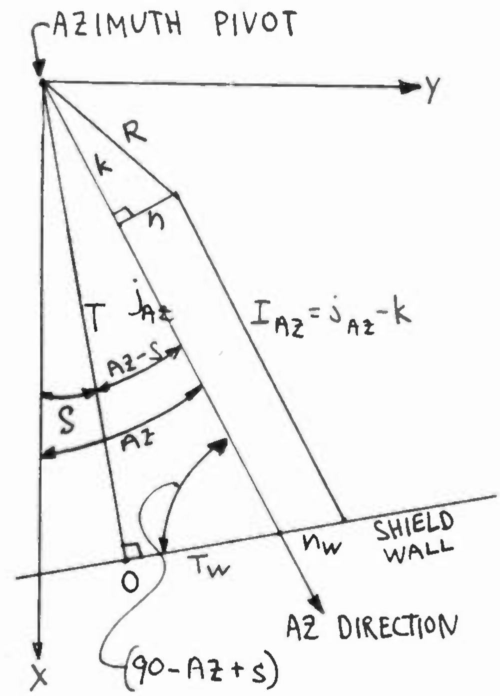


FIGURE 15

SHIELD WALL GEOMETRY, PLAN VIEW

The location of where lines drawn from the reflector edge to the satellite pass over a close-in shield wall is shown.

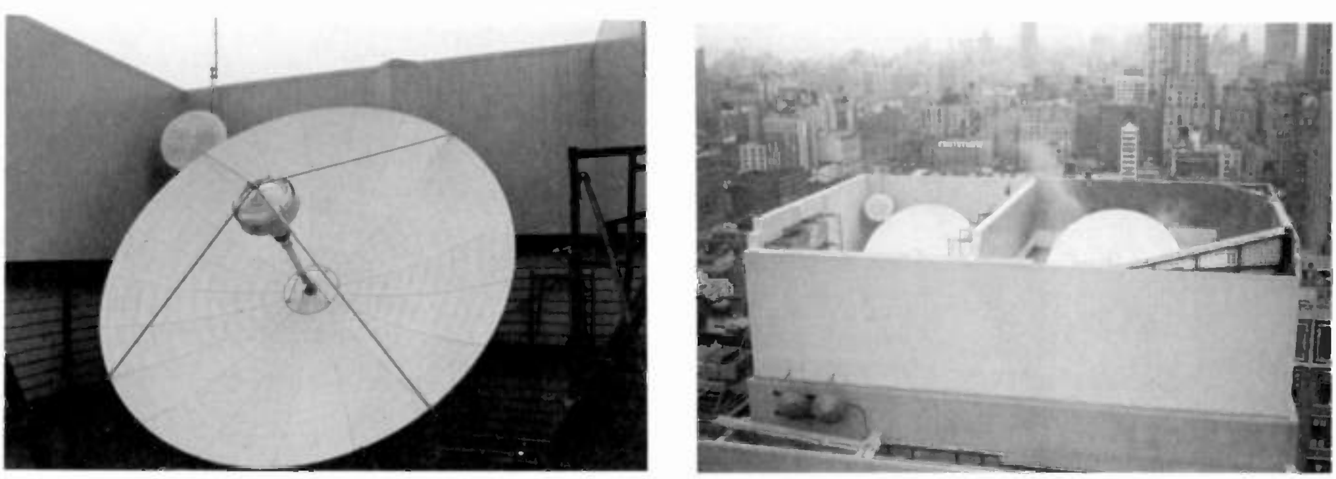


FIGURE 14

ABC NEW YORK CITY UPLINK

Two 9.1 meter C-Band uplink dishes have been placed on the roof of ABC's new technical center in Manhatten. Frequency coordination with terrestrial users required that walls 31 to 36 feet high be placed around the antennas to form a box 60 x 100 feet. Microwave absorber was applied to the inside of the walls, which are as close as six feet to the dishes.

CONCLUSION

Geometric modeling and computer plotting can speed up calculations on earth station siting to the point where the designer is free to play what if games and approach an optimal solution within a reasonable period of time. These techniques are particularly useful for multi-antenna sites and sites with obstructions. They are essential for the type of close-in shield wall built around the ABC New York City uplink.

APPENDIX

The formulas for the calculations presented in this paper are listed in this appendix.

Antenna Geometry

AZ - Azimuth, given
EL - Elevation, given

From figure 4

EPD - Distance from azimuth pivot to elevation pivot, given
EPD_z - given

From figure 5

$EPD_x = EPD \cos(AZ)$
$EPD_y = EPD \sin(AZ)$

From figure 6

A - given
v - given
$MD_{az} = A \cos(EL + k)$
$MD_z = A \sin(EL + v)$

From figure 5

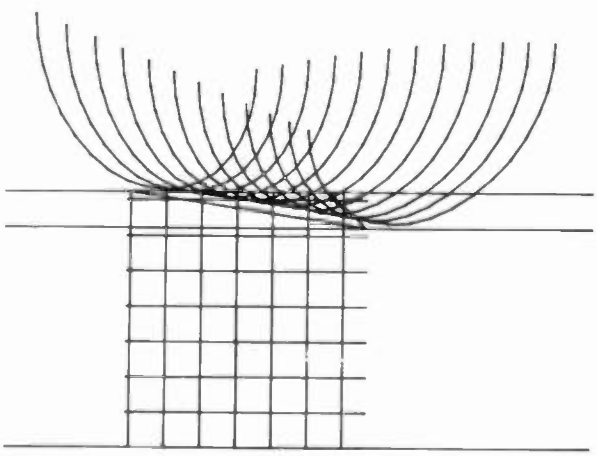
$MD_x = MD_{az} \cos(AZ)$
$MD_y = MD_{az} \sin(AZ)$

Location of M

$(EPD_x + MD_x, EPD_y + MD_y, EPD_z + MD_z)$

From figure 7

R - given
q - given
$k = R \cos(q)$
$n = R \sin(q)$
$k_z = k \cos(EL)$
$k_{az} = k \sin(EL)$
$k_x = k_{az} \cos(AZ)$
$k_y = k_{az} \sin(AZ)$
$n_x = n \sin(AZ)$
$n_y = n \cos(AZ)$
$n_z = 0$



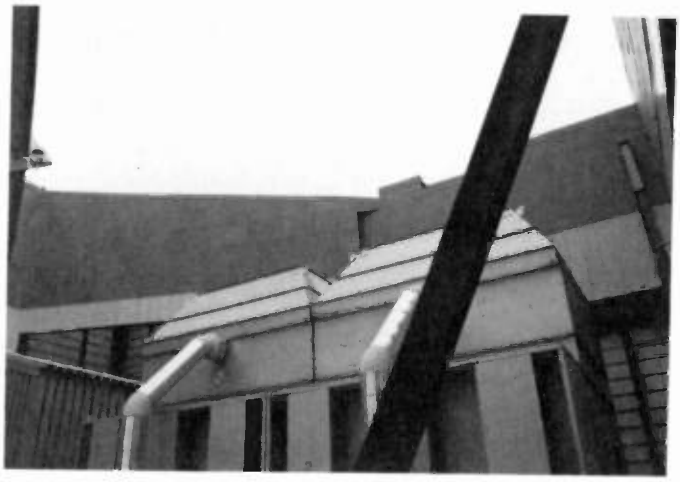


FIGURE 16

SHIELD WALL PLOT AND PHOTOGRAPH

Figure 16a shows a computer generated plot showing a front view of a shield wall. The arcs above the wall show the sections through the beams to various satellites. The sections are cut in the plane of the wall. This plot is the result of successive iterations of shield wall location, height, and shape. Figure 16b shows the angled wall section as built. It has been partially covered with microwave absorber. The back of this section is visible in figure 14 on the right side of the box.

Location of E

$$(EPD_x + MD_x + k_x + n_x,\ EPD_y + MD_y + k_y - n_y,\ EPD_z + MD_z - k_z)$$

Beam Geometry

From figure 9

I - arbitrary

$$I_x = I \cos(AZ)$$
$$I_y = I \sin(AZ)$$
$$I_z = I \tan(EL)$$

Shields

From figure 15

s - given

T - given

$$j_{az} = T/\cos(AZ-s)$$
$$I_{az} = j_{az} - k$$
$$Tw = T \tan(AZ-s)$$
$$(I_{az})_z = I_{az} \tan(EL)$$

Location of F

$$F_z = EPD_z + MD_z - k_z + I_z$$
$$F_w = T_w + n_w$$

TECHNICAL TRADE-OFFS IN DESIGNING SYSTEMS FOR GATHERING NEWS VIA SATELLITE

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Minneapolis, Minnesota

ABSTRACT

This paper examines the various design trade-offs that have resulted in the present-day Conus SNG® system. I will discuss how we arrived at the video and communications transmit formats that are presently employed as well as present a brief discussion of why these particular methods were chosen.

CONUS OBJECTIVES

Conus Communications was formed as a partnership of broadcasters with the objective of making satellite news gathering an effective and affordable addition to the local television broadcasters compliment of news gathering tools. Realizing that satellite news gathering would be a very expensive undertaking for local television stations, Conus sought to minimize the cost of the expensive resources by having a large group of stations share in that cost. Conus also sought to realize a maximum return on the stations' still rather large investment in mobile uplink hardware by allowing that hardware to be used by a large number of entities.

A control center designed expressly for satellite news gathering and staffed by television news personnel provides us with a unique method of reducing the high cost of control center facilities and satellite time. Since this system is designed specifically for TV news, it enables Conus to operate with high speed and efficiency. Stations buy only the satellite time actually required, allowing more stations to use the same satellite and control room facilities. This is required since satellite transponder availability is a poor match to the high peak demand required by television new broadcasts.

It would be impractical for a station with a mobile uplink to routinely send its unit thousands of miles from home, since the travel time would negate the advantages of the rapid satellite delivery. Consequently, each station's strongest use of its mobile uplink is to extend its live news gathering capability in its own region. The Conus system, then, allows that station to use the uplink of another station outside of its own service area. This functionally has the impact of buying a part of fifty uplinks instead of buying just one.

The overall Conus philosophy can be summarized by saying that we are in business to make satellite news gathering work for our member stations. Part of this means that our engineering efforts are directed toward making the mobile uplink unit as cost effective, reliable, and as easy to use as is possible.

VIDEO SYSTEM DESIGN

Since transponder availability and news demand during peak hours are a poor match, two videos per transponder was mandated right from the beginning for business reasons. Since video signal to noise ratios of 52db were obtainable using ordinary frequency division techniques, this method was adopted over the considerably more complicated time division multiplex method which operationally produced few benefits, yet had considerable cost and complexity implications.

Satellite Selection:

When Conus began operation there were no satellite news gathering systems, and satellite availability was wide open. In choosing a satellite system for our initial start-up we examined a variety of systems, finally settling on the SBS satellite series, even though it had the narrowest bandwidth. The narrow bandwidth would mean a slight reduction in video deviation and resultant drop in the video signal to noise, but was more than made up for by the SBS series being able to provide higher EIRP and a resultant higher C/N component in the downlink S/N equation. This meant that the SBS system would provide about the same video performance as other systems, but with slightly higher fade margins. A side benefit of designing for the slightly narrower system resulted in Conus operating a frequency plan that is now capable of fitting into any North American Ku-band domsat without changes in operating parameters.

Video Bandwidth:

Considerable testing was done at SBS's Clarksburg station to arrive at final video system specifications. Dual video transmission was poked, prodded, abused, and maligned in every manner that we could reasonably expect to see occur in the field in order to determine what modulation values and audio subcarriers would perform the best and show minimal degradation in the presence of errant transmissions. The biggest crosstalk problem turned out not to be the transponder, as most of the literature had suggested it would be (mostly based on previous C-band work), but was instead based on actual spectral overlap of the two video signals. This resulted in our selection of the carrier spacing of 24MHz and a receive IF filter response of 20MHz (see Figure 1). This allows for reasonable filter designs to reject the adjacent video signal. In our tests, the only way that we could cause the transponder to produce visible video cross talk was by employing too narrow an upconverter band pass filter, thus creating amplitude modulation on one of the two video carriers. After AM/PM conversion due to the TWT, this would produce visible crosstalk in the adjacent video. This problem is avoided by employing a 24MHz upconverter filter (slightly wider than the required bandwidth) which under normal circumstances serves almost no function other than to eliminate the risk of gross overmodulation or non-standard inputs to the video exciter.

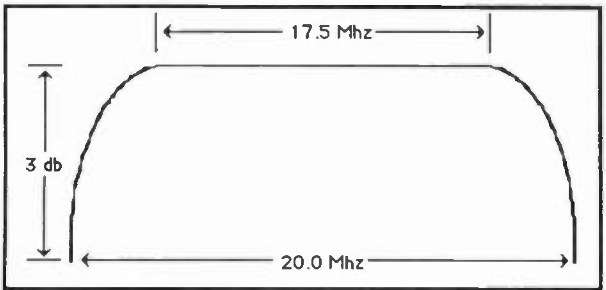


Figure #1 - Receive IF Filter Response
Half Transponder Video

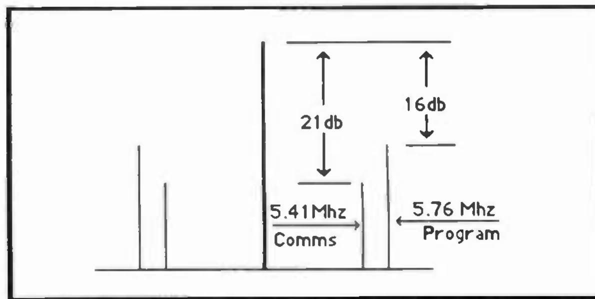


Figure #3 - Spectrum Analyzer Display of
Audio Subcarriers with no Video Modulation Present

Audio Channels:

The addition of audio subcarriers at 6.2 and 6.8MHz caused difficulty with this transmission format since they considerably widened the overall RF channel. Since reducing the video deviation to accomodate these audios was an undesirable solution to this difficulty, Conus decided to lower the frequency of the audio subcarriers instead. The high frequencies of 6.2 and 6.8 are largely a throwback to the days when good quality filters for operation at lower frequencies were not available. Consequently, lowering the audio frequencies to 5.41 and 5.76MHz in present day hardware has essentially no impact on the performance of the audio channels, as well as no impact on the performance of the video channels. Since the FM noise curve increases with frequency, we can use less subcarrier injection at these lower frequencies and maintain the same level of performance (see Figure 2). Reduction of the injection level of these high frequency components into a pre-emphasized FM transmission system has a dramatic impact in reducing the overall RF bandwidth of the channel.

Since the frequency in absolute terms is also considerably lower, there is a second sizable reduction in the size of the RF channel required. These changes allowed us to maintain a high video performance level without sacrifice in either the audio or the video performance standards.

The 5.41MHz subcarrier in the Conus system conveys communications from the mobile uplink to the Conus control center and is always received on a large aperture receive system, we can reduce the injection level of this signal to further reduce the overall RF bandwidth while maintaining the necessary communications-grade audio (see Figure 3). 5.41MHz was chosen for communications because of the greater likelihood of crosstalk from video, and its lower injection level would in turn be less likely to produce audio crosstalk into video in marginal video receive equipment.

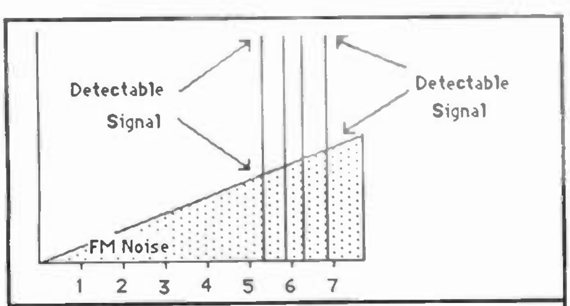


Figure #2 - SC frequency versus FM noise

COMMUNICATIONS SYSTEM DESIGN

Coming from a broadcast background we realized from the beginning that communications support is absolutely vital to the success of any live news gathering effort. We further realize that our success or failure as a news organization may well depend on the effectiveness of our communications system. As amazing as video from the middle of nowhere might seem, providing the communications support is lot tougher. Since Conus is not a satellite carrier, and went into business with an untested business concept, we did not have the luxury in the design of our communications system of being able to even consider putting communications in a separate transponder dedicated to communications. Thus, communications had to share the same transponder with two videos.

Now that we have grown and expanded our operations to multiple transponders we have realized a side benefit from this original plan. Our system is satellite-agile. We can drop our entire communications system and video transmission format into any Ku-band domsat on an occasional use basis. So, as our system grows, we can help meet peak demand by buying excess capacity from other carriers.

Use of a separate transponder also has technical drawbacks. To transmit to a separate transponder, an SCPC type of system must be employed that either employs a separate low-power transmitter diplexed with the video transmitter, which is expensive, or it must add an SCPC signal to the video transmit amplifier requiring the video transmit chain to be backed off from its maximum output power, and even then intermodulation products will fall into a third transponder, or be radiated out of band. It is also not reasonable to expect that a backoff will be properly maintained in mobile environment that is frequently power limited to begin with.

Identifying Required Services

To determine what communications functions would be required, we studied existing ENG systems, and remote television broadcast facilities. From our study of existing ENG systems we were able to determine realistic goals for our satellite delivered communications system.

It should be capable of delivering the same or better service as a well-equipped ENG unit. It would be impractical and contrary to our policy of cost-effectiveness to provide satellite-delivered communications of sufficient magnitude to meet every conceivable remote pickup situation, so what we sought to do was to provide a system that would be easy to use, reliable, and would meet the majority of television news applications. This meant that special event situations requiring excess communications beyond the normal system would have to be met with either add-on satellite equipment brought in for the occasion, or conventional terrestrial based systems. Our service definition studies revealed that we should provide the following communications services to the satellite news gathering unit:

IFB: Interruptable feedback from the stations' in-studio talent would be required to allow the field reporter to converse back and forth with the anchors in the studio. This service was deemed to be the most important one identified.

Private Line (PL): This would be a duplex circuit that would allow the operator or the director in the mobile unit to speak with the director or producer at the station. This circuit effectively extends the station's telephone or headset system into the mobile unit. This circuit also provided an effective point for the Conus operator to monitor and convey operational messages or troubleshooting information to either the mobile unit or station.

The two services above need to be dedicated to a particular station. These would be services that come as a part of the purchase of video transponder time. We also identified a requirement for the mobile unit to be able to talk to the station and to Conus throughout the day in order to conduct the miscellaneous business that news gathering requires.

Satellite Two-Way Radio: This service would allow a unit that is not transmitting video to have a shared access, similar to a conventional two-way repeater, to a two-way radio that talks to Conus, to other units on the same channel, and by telephone interconnection to the home station.

Data: With the advent of more and more stations using small computers in the field to transmit script information and other story information, we saw it as desirable to include a method of transmitting data over the system that, beyond the station's requirements, could also send out Conus schedule information and other system operating notes, thus providing the mobile unit with valuable information as soon as it had acquired the satellite. This sort of data channel would reduce the demand on system voice circuit capacity, and speed up other operations.

TO PHONE OR NOT TO PHONE

All of the above services can be provided to the truck by simply providing conventional dial phone services. This approach, while simple on the surface, drives the cost of the mobile unit communications up rather dramatically. Aside from the cost and reliability implications of adding the necessary signaling circuits to make phones work in a truck, many of the services that are required are simply not bi-directional circuits, such as IFB. Further, it does not make good sense to burden the truck operator with connecting services such as IFB. It's normally desirable to have IFB connected ahead of time, but if this connection is all the way through to the truck ahead of time, then the overall system circuit count is increased by that circuit.

In the Conus system, IFB's are connected ahead of time, but they are connected to the switch gear at Conus ahead of time, where the cost of having that call on hold and available for immediate use is small and the crew in the field has not spent any time or energy making that connection. Once the circuit is required, it is switched through. In the case of a mobile uplink doing reports for more than one station, this approach lends itself to switching IFB's from one station to the next as quickly as you can push a button. This technique would be difficult in a truck where the truck itself has to dial the IFB lines individually.

So we adopted a strategy of putting the phone lines on hold where you have the room and the staff available to do that without consuming the time and energy of the field crew, or burdening the truck with the additional expense of more communications hardware. This approach had the side advantage that a station phoning its truck is calling the Conus control center and always gets an answer whether or not the truck is operational. Even when the truck is not operational, Conus frequently is aware of the exact status of the truck, and that information can be passed on. Further, many calls from the truck do not go to the station, but rather to Conus to verify schedule information and transponder assignments.

After several years of operational experience, we have discovered that conventional phones in the mobile unit are frequently just too slow.

COMMUNICATIONS SYSTEMS DESIGN TRADE-OFFS

The results of our studies told us that we should provide these outbound services: IFB, PL, two-way radio, and data. This block of services is associated with each video carrier. We identified these inbound services that would be required: PL, two-way, and data.

Having to fit communications into a transponder 44MHz wide that already has two videos in it that produce saturation is a demanding task. To accomplish this we had to choose an approach that minimized the use of RF bandwidth, and minimized the number of individual RF carriers within the transponder.

Analog or Digital?

Digital transmission was examined briefly, but was rejected for a variety of reasons. Digital transmission, while frequently employing less RF power (not a major concern) tends to occupy more bandwidth than its analog counterpart. These transmissions also do not fade as gracefully as analog transmissions. The human ear has a remarkable power to distinguish between noise and the desired signal. This free processing capacity did not exist in digital systems, and when added to the increased cost and complexity of digital systems, at least in the beginning, led to our adoption of an analog approach.

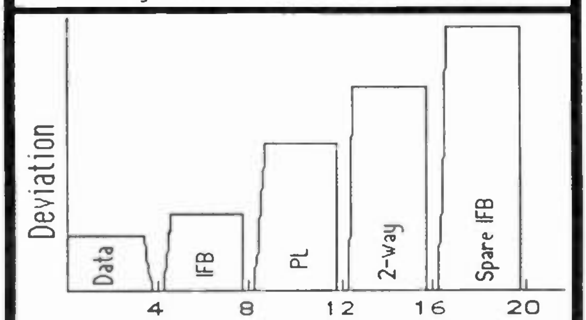
SCPC or MCPC?

When we started examining the signals we would be sending to our mobile units, SCPC (single channel per carrier) was the logical choice, but as we examined the total bandwidth reqirement of all these different SCPC channels, we realized that we were running into a bandwidth limitation. Furthermore, the close spacing and narrow channel requirements would require a higher frequency stability and corresponding higher cost of the downconversion chain in the mobile unit. This ran counter to our philosophy of minimizing the cost to the individual station, and concentrating as much of the high stability high tech elements of the system in the control center so that no one station would be burdened with the cost. The problems with the SCPC approach led us to explore the less common MCPC (multiple channel per carrier) method that would allow us to put up to five voice grade circuits in the space normally occupied by one program grade SCPC (see Figure 4). This wider, more powerful carrier would relax the frequency stability requirement in the mobile unit with the attendant reduction in cost and complexity. This approach also had the benefit of putting fewer RF carriers in the transponder and conserving total transponder

RF bandwidth. This also results in a more manageable intermodulation scenario. MCPC had the side benefit that all services associated with a particular video channel would switch when the demodulator frequency for the MCPC channel was switched. The use of MCPC also tends to randomize the modulation of the RF carrier reducing the energy density. This has recently become important as the FCC now bases power allocation on the basis of energy density into the antenna (as well as maximum spacecraft EIRP).

Even though one MCPC carrier is devoted to one video carrier, we chose to put both the outbound MCPC channels at the very lower edge of the transponder. This allowed us the option of operating the transponder with either center channel traffic or dual video traffic. This permits us to resell transponder capacity during off-peak news hours to non-news customers, thus again helping to reduce the cost of news operations to our stations without having to shut down any of the communications capacity due to the outside customers.

Figure #4 - MCPC Channel



Inbound Communications

Subcarrier: One of the obvious routes for inbound communications from the mobile unit to Conus was the use of subcarriers. Subcarriers are available to us at virtually no cost in either dollars or performance, so we adopted an approach that a unit that is on the air transmitting video would transmit all of its communications on one subcarrier. In the Conus system the communications subcarrier carries the inbound half of the duplex private line channel, as well as a multiplexed data channel.

An additional advantage in using subcarriers for inbound communications from mobile units sourcing video is that the communications system can be implemented in a phased-in approach where the simplest system employs only a communications receiver and the ability to talk to Conus on the communications subcarrier. In this scenario, Conus on a one-way basis, will issue authority for a truck to initiate a transmission at low power. Upon seeing the truck responding with a low power video carrier, Conus knows that the truck is in position, ready, and hearing our transmissions. Conus then authorizes the unit to increase the power to the proper video operation point whereupon we can converse fully and carry on the business that any other fully equipped Conus mobile unit could undertake. This approach has allowed us to implement our communications system early and at low cost without impacting the overall system design, or delay start-up waiting for the long lead items.

The use of the subcarrier for inbound communications from mobile units producing video allows us to avoid asking either an upconverter or an HPA to amplify more than one RF carrier at a time. The technique of sending multiple carriers through an HPA is fine if the unit is operating below saturation, but we did not believe this to be a realistic assumption in a mobile unit with a small antenna and limited uplink power to begin with. The quest for the last little bit of video performance would frequently demand that the amplifier be driven to the point where the additional communications carrier would produce an intermodulation product on the other side of the video carrier of unacceptably high amplitude. Actual field testing in Conus-type units verified these fears. By not using SCPCs for communications with trucks sourcing video, it enabled us to make SCPC space available for those trucks not sourcing video, increasing our total communications availability, and eliminating a potential interference source.

SCPC: A study of various manufacturers' video exciters revealed that all the units we tested had upconverters that were sufficiently low in phase noise and frequency drift that they could be employed to upconvert narrow band SCPC signals of suitable voice grade quality. This provided us - essentially for free - with our communications grade transmit chain. By adding two inexpensive fixed frequency SCPC modulators we were able to provide our inbound two-way radio and data communications services. The only restriction on these is that they could not operate from a truck sourcing video. This restriction, however, turns out not to be a major problem, since trucks sourcing video have their own exclusive circuits available to them. In this system inbound SCPC traffic has a fixed frequency relationship with its corresponding video channel. These signals are low power, narrow band SCPCs that could not be received on the small truck antenna directly, but work nicely into the large communication receive antenna at Conus Control. At Conus the receive audio is either repeated back out to other units on the MCPC channel, or routed down the phone line in the case of a phone call.

Spectrum: When you put together the two video signals, the two MCPCs that support them, and the inbound SCPCs you will come up with the loading depicted in Figure #5, the Transponder Spectrum. It can take as many as seven different uplinks to create the full load condition shown in Figure #5. As you can see, this represents a rather crowded transponder despite our efforts of minimizing bandwidth and numbers of carriers.

Video signals and SCPC signals frequently share some of the same spectrum. Fortunately, these are dramatically different types of signals and do not normally interfere with each other. The SCPC is simply too weak to produce visible interference to the video, and the video signal moves its energy around at such a high rate that the period of time that video energy conflicts with the narrow band SCPCs is small enough that audible crosstalk from video is rare.

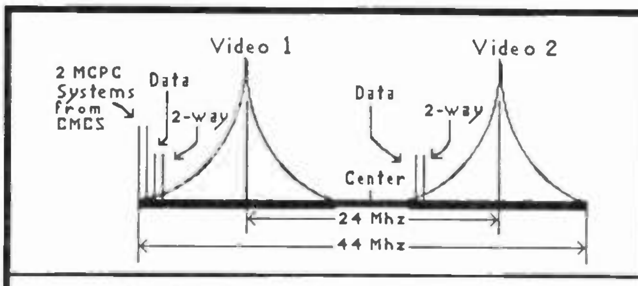


Figure #5 - Transponder Spectrum Fully Loaded

ONE CUSTOM BOX

In order to keep the cost of our communications system within reason, we wanted to use off-the-shelf components and equipment wherever possible. Our resultant design requires only one custom built box unique to the system. It is the SCP-1 two-way radio. This device pulls together all the loose ends and turns it into a unified system. The SCP-1 contains the SCPC modulators that drive the video transmit chain, as well as the associated audio circuits. This unit also has some protection logic, that looks at exciter configuration switches and other signals to prevent errant transmissions when equipment outside the communications system is incorrectly configured.

Part of the SCP-1 logic is designed to control a digital attenuator. While a digital attenuator is not the only method available for power control, it is one of the best. It can be implemented using either a 70MHz or a 14 GHz device, depending on the RF chain. The SCP-1 automatically sets the correct system gain so that video transmissions and SCPC transmissions originate with the correct RF power level. The device also contains the manual power controller that provides the correct startup power for video transmissions, and the graceful increase in power that's required for the Conus power up procedures. Built into the SCPC-1 is considerable expansion capability that allows external equipment to be added to the system, expanding the communications capability. Stations can use this capability to tie existing intercom, two-way radio, and other accessory systems into the satellite communications, thus customizing it to meet their own unique requirements. We have also built in additional access to the RF chain so that entirely separate communications systems can be added to the system while still taking advantage of its protection logic and power control systems.

SUMMARY

Principal features of the Conus SNG® system are low cost, no added microwave plumbing beyond that required for video, modest stability requirements imposed on the up and downconverter chains, a phase-in approach that allows stations to start with a communications receiver only, a system with enough designed-in expandability to be able to accommodate changes in use and improvements in technology. These include the use of solid state comm's transmission, and additional circuit capacity.

DEVELOPMENT OF A KU-BAND 2.3M ANTENNA FOR A TRANSPORTABLE EARTH STATION SYSTEM

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Abstract

The development of a 2.3m offset-fed antenna system is described. The primary factors governing the design of this antenna are the system requirements for a news gathering by satellite system and the stringent radiation pattern specifications imposed by the FCC for transmitting earth station antennas. How these factors influenced the antenna design will be discussed. The resulting antenna system will be described, and data representing the electrical performance of the offset-fed 2.3m antenna system will be presented.

Introduction

During the development of this antenna for a news gathering by satellite application many requirements have shaped the antenna design. These requirements are listed below:

- The antenna's operation must be compatible with the existing Ku-band geostationary satellite systems. Specifically, that the antenna transmit in the 14.0-14.5 GHz frequency band, receive in the 11.70-12.20 GHz frequency band, radiate energy that is linearly polarized with 35 dB of on-axis polarization discrimination and that the polarization be adjustable by motor control.
- The radiation pattern of the antenna must meet the FCC regulation 25.209.
- The antenna must be capable of transmitting and receiving on both polarities simultaneously. That is, the feed system must allow incorporation of a 4-port combining network.
- The antenna gain should equal or exceed 48.5 dBi at 14.25 GHz.
- The antenna gain should be 46.8 dBi at 11.95 GHz.
- The antenna design should allow the mounting of a sizable RF package containing low noise amplifiers and redundant power amplifiers close to the antenna feed terminals. Thereby minimizing transmission line losses to maximize G/T and effective radiated power.
- The antenna system must be operable in winds of 30 mph gusting to 45 mph without significantly compromising performance.

Antenna Size and Geometry

The requirements that the antenna transmitting gain be 48.5 and the physical size limitations of a vehicularly mounted, deployable antenna led to the choice of an antenna size of 2.3m (this requires an antenna efficiency of 63% for the four-port version).

The simplest approach for illuminating a 2.3m antenna aperture is to use a symmetrical, prime-focus antenna geometry. This is the most common of earth station antenna types. In this scheme the feed horn is held by rigid struts or a combination of waveguide bends and guy wires at the focal point of a paraboloid of revolution.

This approach would be well suited to this application if it were not for three of the design requirements which must be met. They are: that the antenna radiation pattern meet the FCC specification 25.209, that the antenna feed system be a four-port device and that the feed system be rotatable by motor control to change and optimize polarization.

The relevant sections of the FCC regulation 25.209 states[1]:

25.209 Antenna Performance Standards

(a) The gain of any antenna to be employed in transmission from an earth station in the fixed-satellite service shall lie below the envelope defined below:

(i) In the plane of the geostationary satellite orbit as it appears at the particular earth station location:

$29 - 25 \log \theta$ dBi	$1° < \theta < 7°$
-8 dBi	$7° < \theta < 9.2°$
$32 - 25 \log \theta$ dBi	$9.2° < \theta < 48°$
-10 dBi	$48° < \theta < 180°$

Where θ is the angle in degrees from the axis of the main lobe and dBi refers to dB relative to an isotropic radiator. For the purposes of this section, the peak gain of an individual sidelobe may not exceed the envelope defined above for θ between 1° and 7°. For θ greater than 7° the envelope may be exceeded by 10% of the sidelobes, but no individual sidelobe may exceed the envelope by more than 3 dB.

(ii) In all other directions:

Outside the main beam the gain of the antenna shall lie below the envelope defined by:

$32 - 25 \log \theta$ dBi	$1° < \theta < 48°$
-10 dBi	$48° < \theta < 180°$

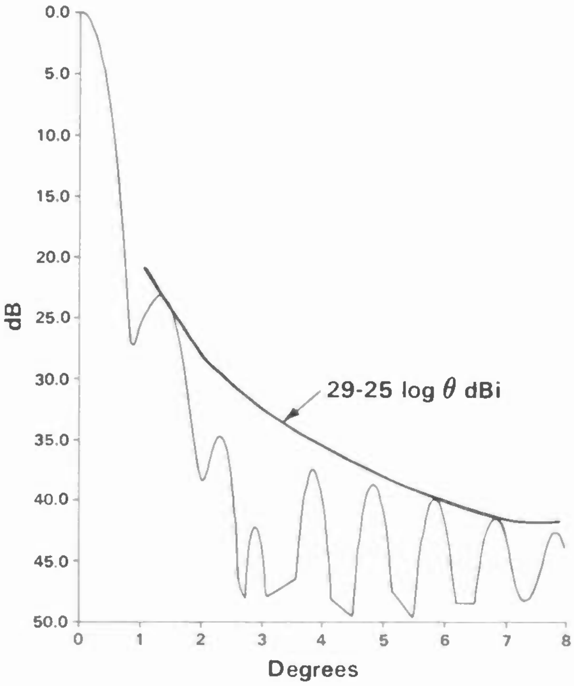
Where θ is the angle in degrees from the axis of the main lobe and dBi refers to dB relative to an isotropic radiator. For the purposes of this section the peak gain of an individual sidelobe may be reduced by averaging its peak level with the peaks of the nearest sidelobes on either side or with the peaks of two nearest sidelobes on either side provided that the level of no individual sidelobe exceeds the gain envelope given above by more than 6 dB.

(b) The off-axis cross-polarization isolation of any antenna to be employed in transmission at frequencies between 5925 and 6425 MHz from an earth station to a space station in the domestic fixed satellite service shall be defined by:

$19 - 25 \log \theta$ dBi	$1.8° < \theta < 7°$
-2 dBi	$7 < \theta < 9.2°$

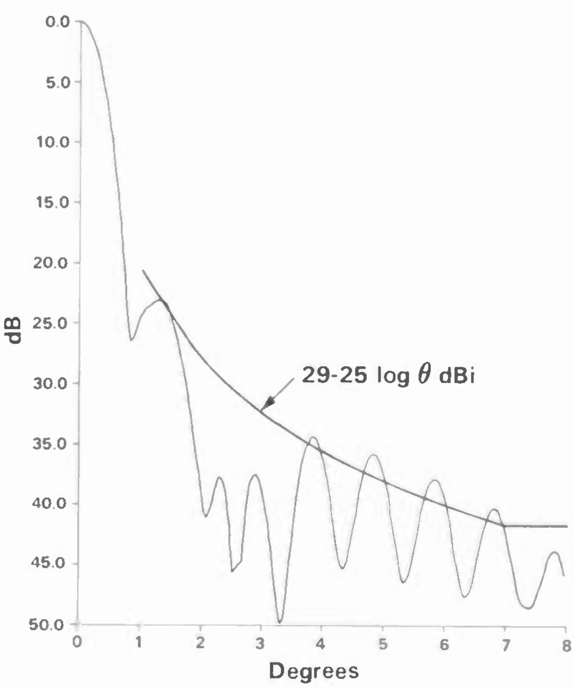
The regulation that the antenna sidelobe levels fall below the 29-25logθ(Theta) curve from one to seven degrees off of the main beam peak in the plane of the satellite arc, is the most demanding of the design requirements. A more complete discussion of earth station antenna sidelobe regulations is presented by A.G. Uyttendaele[2].

The moderate electrical size of the antenna (107 wavelengths in diameter at 14 GHz) causes the sidelobe structure within the first seven degrees to be especially sensitive to aperture blockage, and this sensitivity then becomes a major design consideration. This effect has been investigated by C.M. Knop[3]. Figures 1 and 2 present the predicted radiation patterns of a 2.3m prime-fed symmetrical antenna with central aperture blockage diameters of 4 inches and 6 inches respectively. The pattern envelope in the 4 inch blockage case just meets the FCC specifications, and in the 6 inch case the specification is violated. This theoretical work has been confirmed by measurements on the Andrew antenna pattern test range.



Predicted near-in (0° to 8°) pattern of 2.3m Symmetrical Prime Fed Antenna, 14.25 GHz, 0.014" RMS, 4" blockage

Figure 1

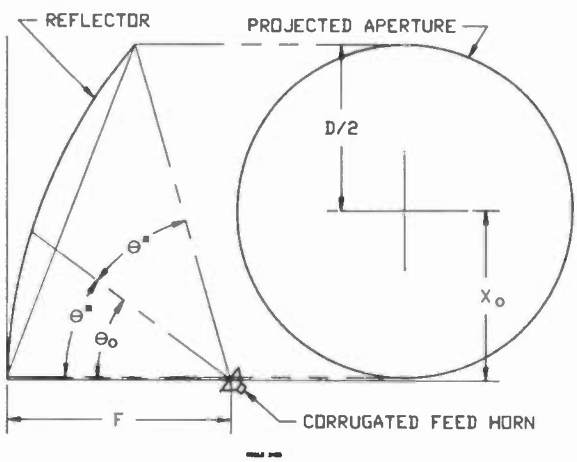


Predicted near-in (0° to 8°) pattern of 2.3m Symmetrical Prime Fed Antenna, 14.25 GHz, 0.014" RMS, 6" blockage

Figure 2

Since the requirement of a four-port feed system with motor rotation requires a relatively large feed system (greater than 6" diameter) an alternative to a symmetric reflector design had to be adopted.

A prime-focus offset-fed geometry was chosen for the final antenna design. This choice effectively eliminated near-in pattern sidelobe problems by removing the feed system blockage from the radiating aperture. This geometry is shown in Fig. 3. The essential difference between an offset-fed and symmetrically fed antenna system is that in the offset-fed case the feed horn is rotated about the focal point in the vertical plane to illuminate a section of a paraboloid rather than a full paraboloid (i.e. rotated by θ_0 in Fig. 3).



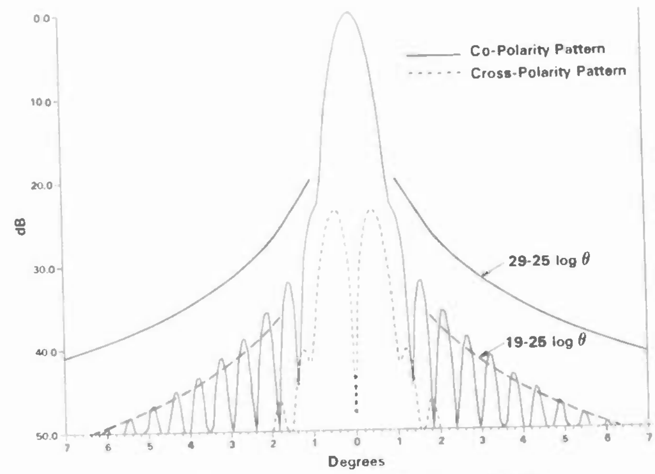
F = 60.00 INCHES
θ_0 = 37.4°
θ^* = 36.7°
$D/2$ = 45.00 INCHES
X_0 = 45.75 INCHES

GEOMETRY OF THE 2.3M OFFSET FEED ANTENNA

Figure 3

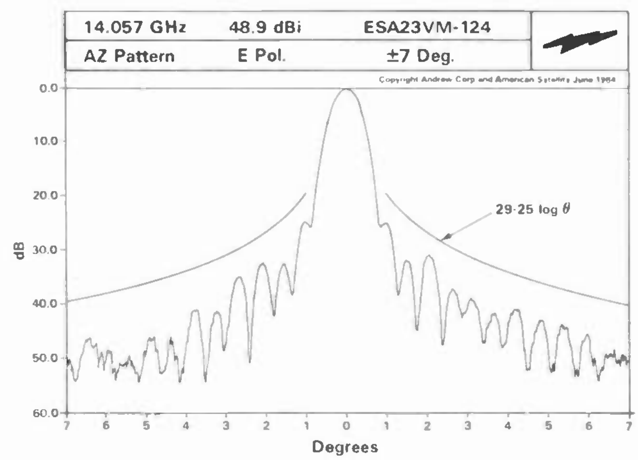
Radiation Pattern Performance

The corrugated feed horn which illuminates the paraboloid section employs an aperture diameter of 1.5 inches. The expected radiation patterns of this antenna system were predicted by C. M. Knop[4] (Fig. 4) and confirmed by satellite and test range measurements (Figs. 5, 6, 7, 8, 9).



Predicted near-in (0° to 7°) pattern of 2.1m offset ESA for ABC (14.25 GHz, E-plane, horizontal plane, 0 RMS)

Figure 4



Radiation Pattern Generated via Satellite Link
Figure 5

Another critical characteristic which effects sidelobe levels in the near-in region of earth station antenna radiation patterns is the deviations of the reflector surface from the theoretical shape. These surface imperfections also reduce the gain of an antenna system. In the case of a Ku-band antenna theoretical and experimental work have shown that these deviations must have a root mean squared (RMS) value of less than .015 inches. Prototype and production versions of this antenna system have been measured for surface deviations and have consistently met the 0.15 inch RMS criterion[5]. Fig. 10 is contour plot of a typical production unit (2.3m offset reflector).

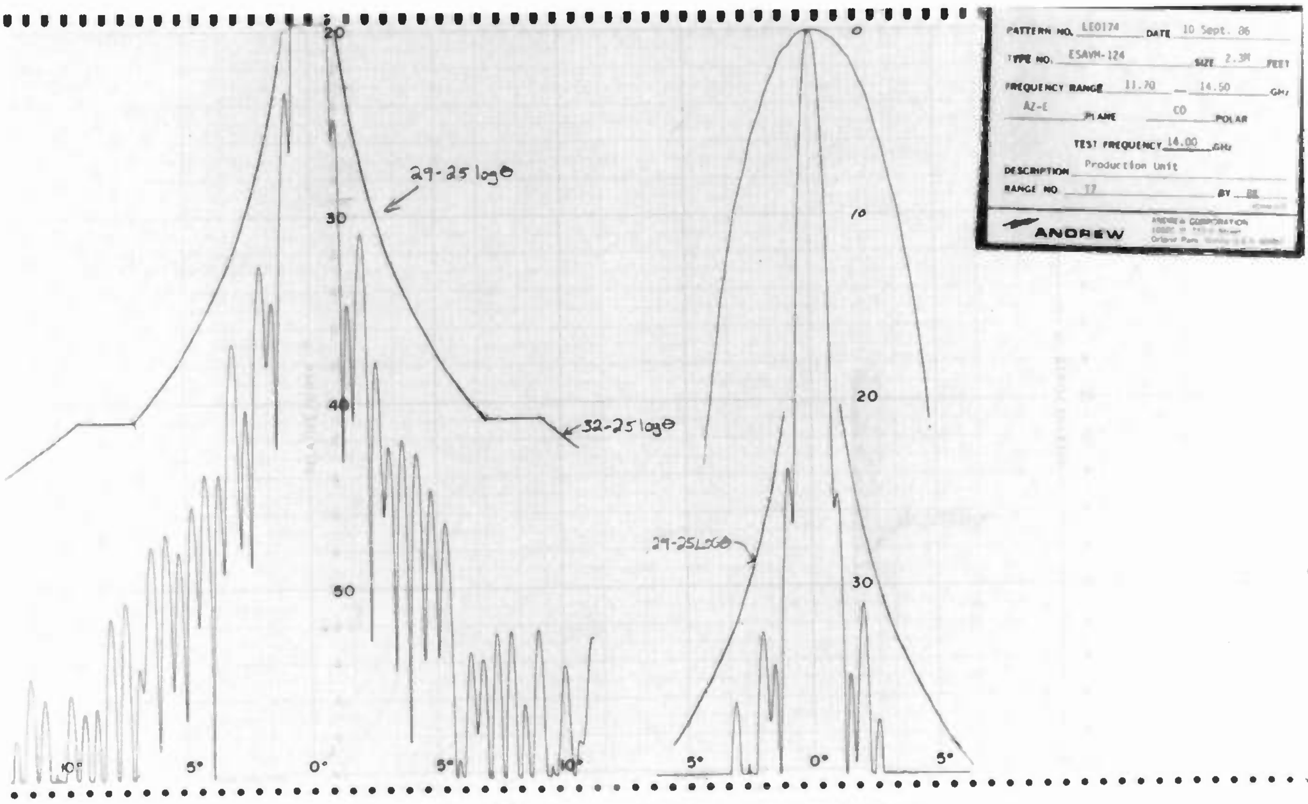


Figure 6 - 14.0 GHz Co-polar Radiation Pattern

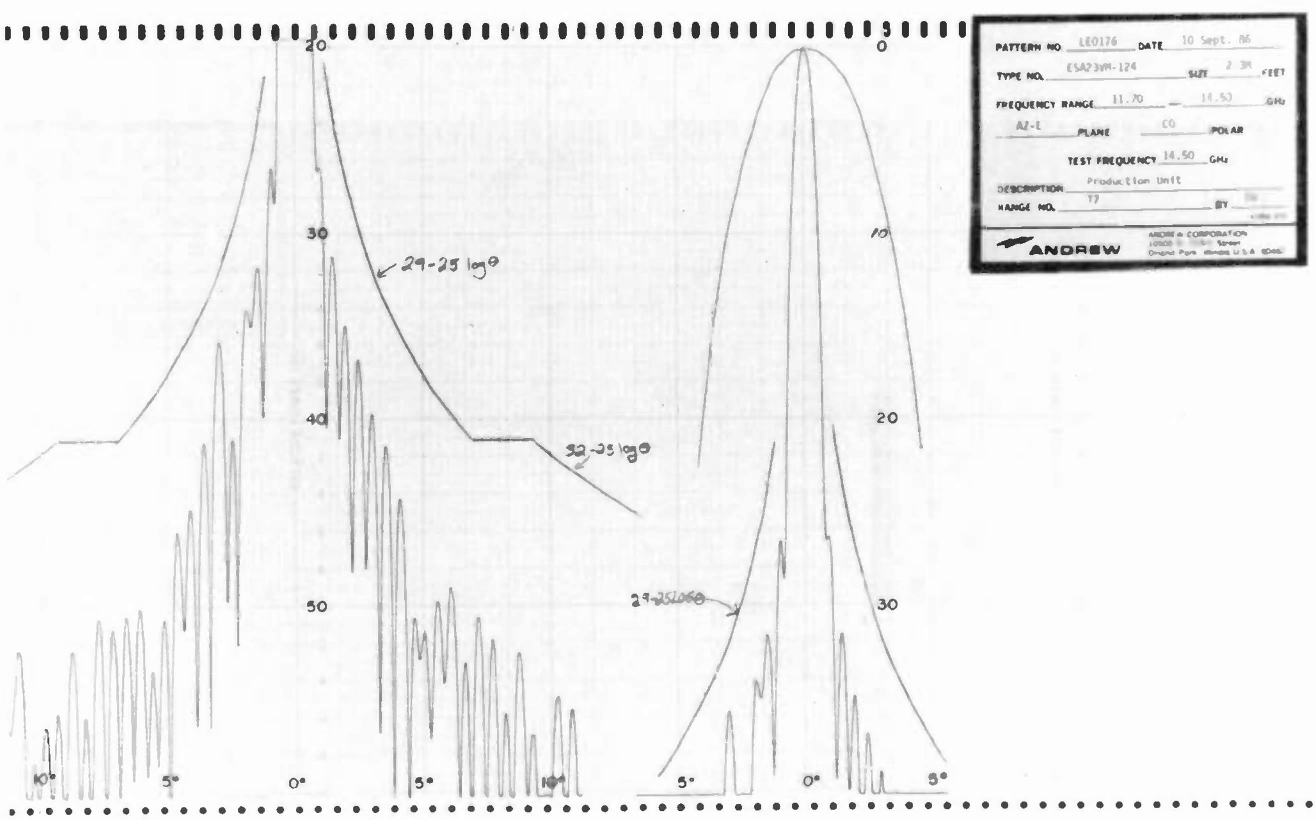


Figure 7 - 14.50 GHz Co-polar Radiation Pattern

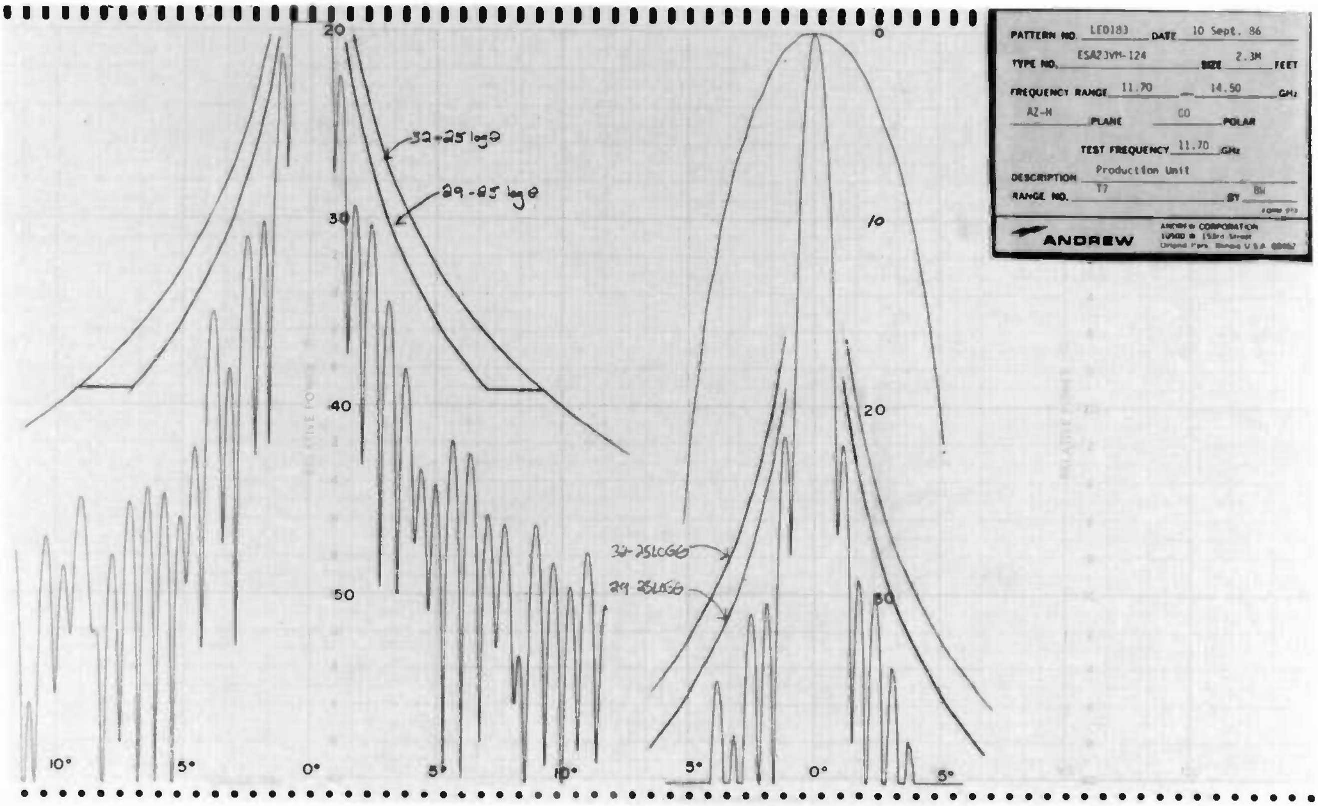


Figure 8 - 11.70 GHz Co-polar Radiation Pattern

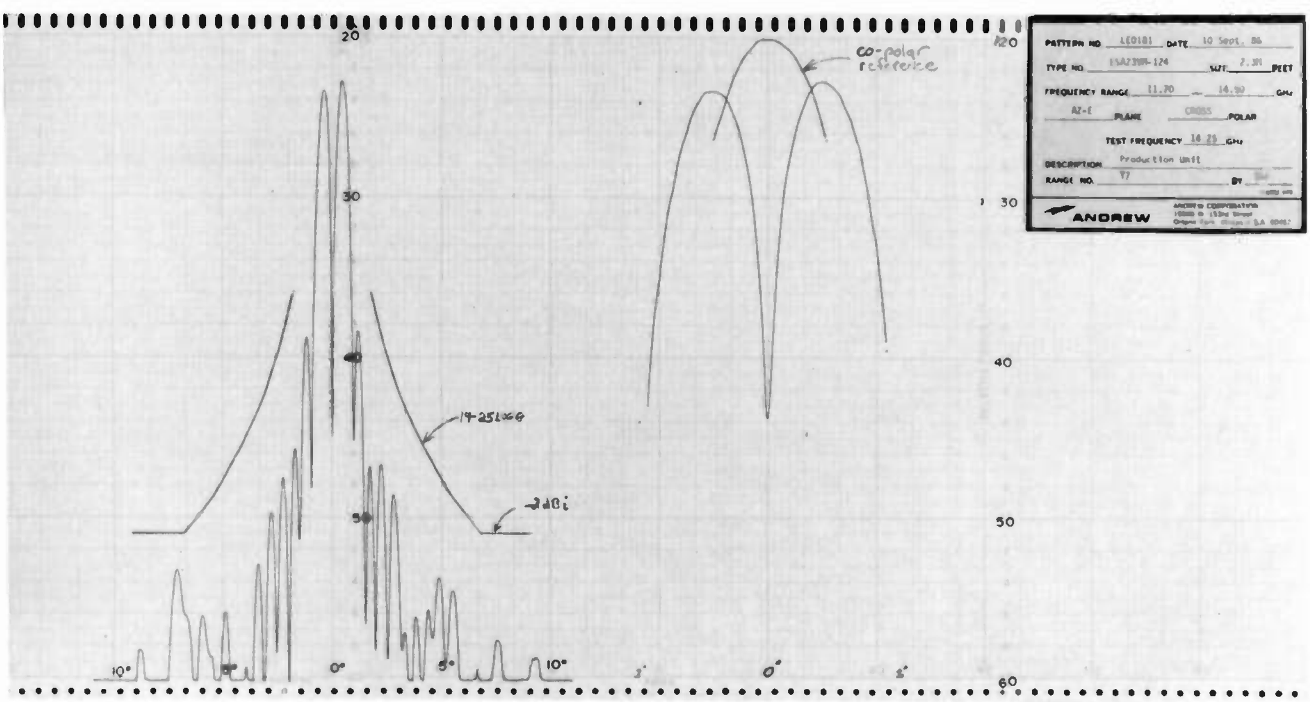
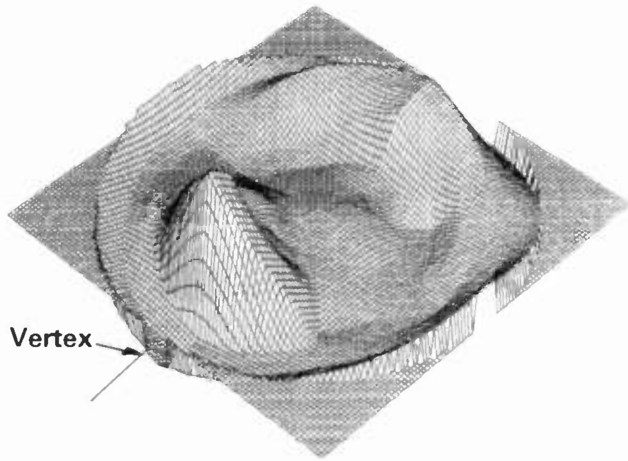


Figure 9 - 14.25 GHz Cross-polar Radiation Pattern



Deviation

Positive Maximum = 0.032 inches
Negative Maximum = 0.015 inches
RMS = 0.008 inches

Contour Plot of 2.3m Reflector Surface
Figure 10

The Antenna Gain and Feed System

The four-port feed system developed for this antenna is pictured in Figs. 11 and 12. The system consists of a corrugated feed horn, an orthomode transducer, two 18-inch sections of WR75 flexible waveguide, two 8-inch sections of WR75 rigid waveguide, and two diplexing arms. The gain budget for both the two-port and four-port versions of this antenna system is presented in Table 1. (The two-port version eliminates the two diplexing arms.)

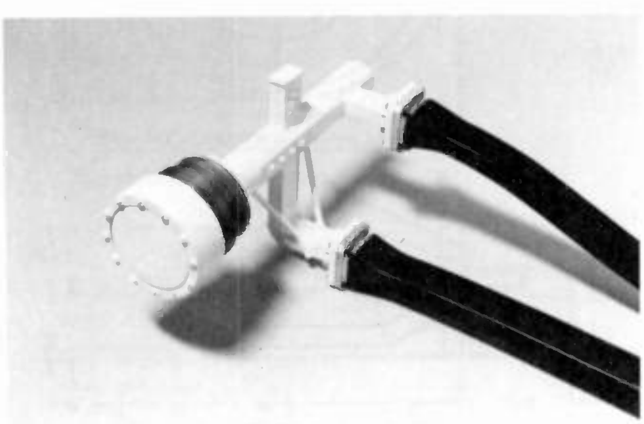


Figure 11 - Feed Horn, Orthomode Transducer and WR75 Flex-twist Waveguide

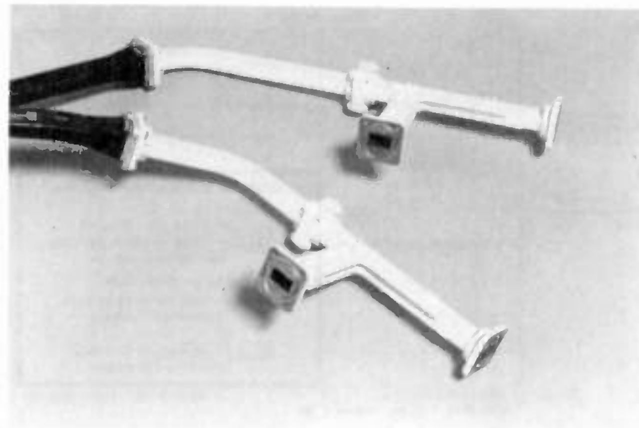
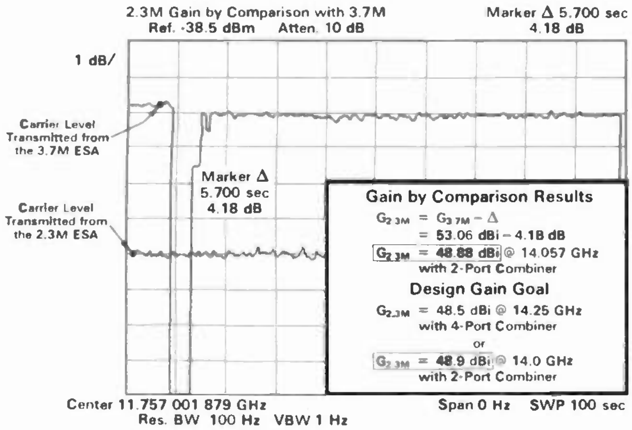


Figure 12 - Diplexing Arms and WR75 Waveguide

TABLE 2 - Antenna System Gain Budget

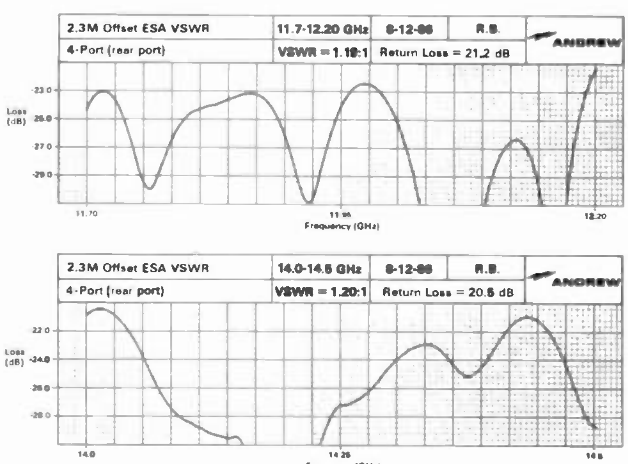
	11.95 GHz	14.25 GHz
Antenna Directivity (Determined from Range Pattern Integrations)	47.60 dBi	49.37 dBi
O.M.T. (2-port) Losses	.12 dB	.10 dB
Corrugated Horn Losses	.04 dB	.03 dB
Transmission Losses (VSWR = 1.30:1)	.08 dB	.08 dB
Cross-polarity Power Losses	.06 dB	.06 dB
Antenna Gain (2-port)	47.30 dBi	49.10 dBi
Additional 4-port Feed Losses Including Diplexing Arms	.26 dB	.26 dB
18" of Flex-twist Waveguide Insertion Losses	.18 dB	.18 dB
8" of WR-75 Waveguide Insertion Losses	.06 dB	.06 dB
Antenna Gain (4-port)	46.80 dBi	48.60 dBi

The antenna gain was confirmed by a transmit gain by comparison measurement. This test was performed over a satellite link using a calibrated 3.7m dual-reflector antenna as a gain standard for comparison. The gain was measured to be 48.88 dBi at 14.057 GHz as depicted in Fig. 13. The test was performed on a two-port version of the 2.3m antenna at 14.057 GHz. Consequently the expected transmit gain should be 48.90 dBi. (There is a 0.2 dBi difference in this expected gain value from that gain value given in Table 1 due to the measurement frequency of the gain by comparison test.)



Gain by Comparison
Figure 13

Figure 14 presents test results of VSWR measurements on the antenna system. These results demonstrate that the antenna system easily meets its VSWR specification of 1.3:1.



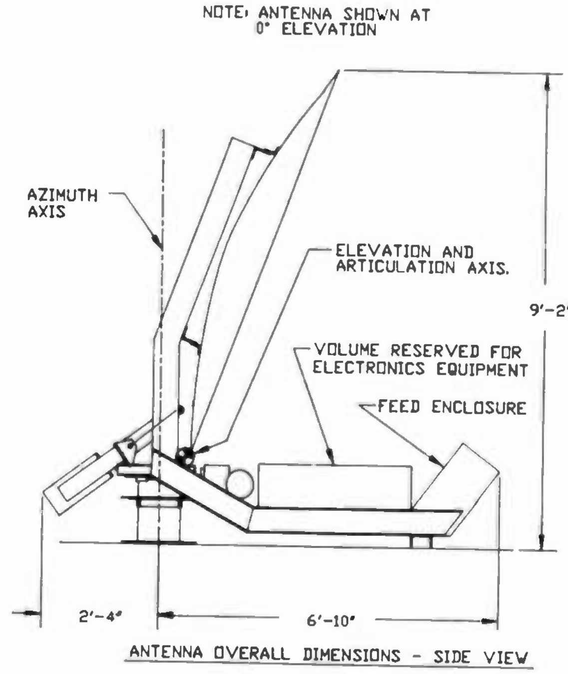
Antenna VSWR
Figure 14

Mechanical Description

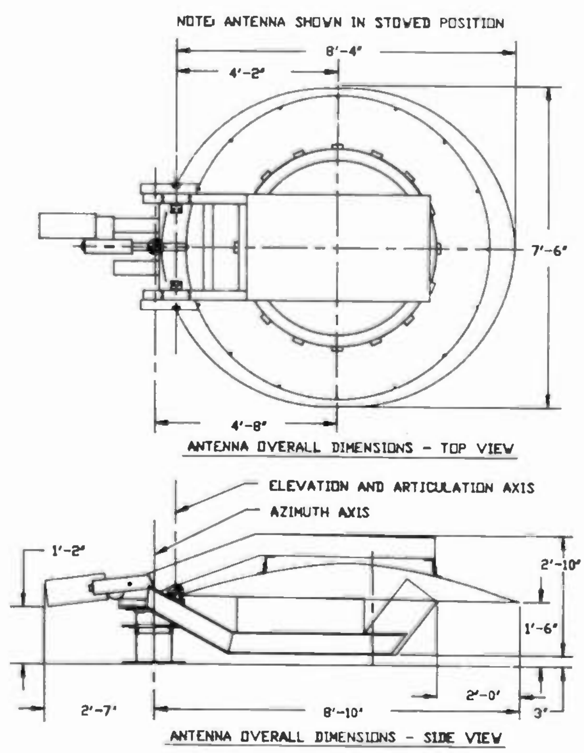
The offset-fed antenna configuration allows the placement of the electronics package very near the input to the four-port feed system. This configuration also provides for a transportable earth station that is quite compact in its stowed position. Figures 15 and 16 show the antenna system in its deployed and stowed positions respectively.

For a more complete mechanical description of the antenna system refer to "2.3 Metre Offset Antenna System for News Gathering by Satellite" [5].

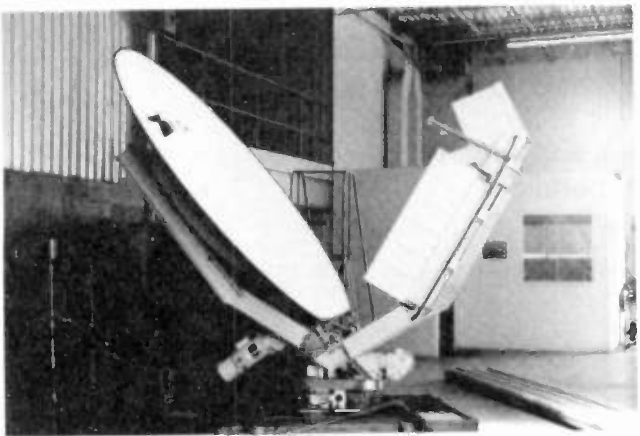
Figure 17 pictures a production version of a 2.3m offset-fed earth station antenna ready for vehicle mounting.



2.3m Offset ESA Deployed
Figure 15



2.3m Offset ESA Stowed
Figure 16



2.3m Offset ESA
Figure 17

Conclusion

An antenna system has been described that meets or exceeds all of the goals established before its development. In particular the 2.3m offset design selected provides radiation patterns that comfortably meet the FCC's stringent 2 degree pattern specification while simultaneously exhibiting high gain and efficiency. The antenna system performs well with either a two-port (orthogonal transmit and receive ports) or a four-port (dual polarized transmit and dual polarized receive ports) feed network and allows the integral mounting of a large electronics package immediately adjacent to the feed system. Finally, motorized polarization, azimuth, and elevation drive systems are provided which are well suited to this mobile satellite news gathering application.

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[1] Federal Register/Vol. 48 No. 173/Tuesday, September 6, 1983/Rules and Regulations.

[2] A. G. Uyttendaele, "Evolution of Antenna Sidelobe Regulation", IEEE Transactions on Broadcasting, Vol. BC-32, No. 4, p. 85, December, 1986.

[3] C. M. Knop, "Inadequate Performance of Andrew's 2.3m Ku-band Symmetrical Prime-fed Antenna as a Four-port at the 14 GHz Band", Andrew Corporation internal memorandum, December 8, 1986.

[4] C. M. Knop, "2.1m 12/14 GHz Offset ESA for ABC", Andrew Corporation internal memorandum, February 12, 1986.

[5] T. H. Hamilton, "2.3m Theodolite Analysis", Andrew Corporation internal memorandum, September 10, 1986.

[6] T. E. Charlton, K. E. Linehan, D. B. Mowry, "2.3 Metre Offset Antenna System for News Gathering By Satellite", IEEE Transactions on Broadcasting, Vol. BC-32, No. 4, pp. 96, December, 1986.

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The authors would like to acknowledge the efforts of the many fine design and manufacturing engineers, technicians, and draftsmen whose talents and hard work are responsible for the successful development of this product. We would also like to thank those who helped in the preparation of this paper.

Finally, the authors would like to recognize C. M. Knop, Andrew Corporation Chief Scientist and his associate, Y. B. Cheng, for their theoretical design work on the 2.3m offset-fed ESA.

VOICE TRANSMISSION CONSIDERATIONS FOR SATELLITE NEWS GATHERING OPERATIONS

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ABSTRACT

A valuable asset to the current generation of satellite news gathering networks is the ability to provide automatic connection (i.e., no human intervention) to the public switched telephone network (PSTN) on demand. This capability is now provided by connecting the mobile earth stations in a full mesh, demand assigned multiple access/single channel per carrier (DAMA/SCPC) satellite network, controlled by a central hub which interfaces terrestrially with the PSTN.

Cost, space, and operation constraints, inherent in satellite news gathering vehicles, and the multiple random accesses of demand assigned satellite capacity dictate careful overall transmission system design. The paper presents design and implementation data compiled from a newly operational network on GSTAR II, and provides useful information pertaining to satellite news gathering vehicle design.

INTRODUCTION

The advent of satellite news gathering is bringing new levels of coverage and operational autonomy to local news organizations. The mobile satellite news gathering vehicle has unique requirements for voice and data communications. Ideally, the vehicle should have ready access to the public switched telephone network (PSTN), in order to link producers and directors with on-site news staff. Direct access to the network control center, and to other satellite news gathering vehicles, is also desirable. Several approaches to meet these communication requirements have been used over the past few years, with varying degrees of success.

First Approaches

One of the first solutions to this problem involved voice subcarriers with the video feed. This was a straight forward solution, but generally constrained voice operations to when and what transponder the video feed occurred on.

Later approaches used separate equipment for generation of voice carriers. These carriers would then be placed at the band edge of the transponder. This solution avoids inconveniences associated with using voice subcarriers, but has the disadvantage that the voice carriers are subject to variable intermodulation and small signal suppression from the transponder travelling wave tube amplifier (TWTA) being fully saturated by the video feed.

More recent solutions to the voice communications problem use a separate dedicated transponder for the voice carriers. This allows the satellite transponder to be operated in a near linear region. Many voice carriers can be accommodated, earth station and transponder power levels are stable and intermodulation can be managed to a minor level.

DAMA is often selected to permit large numbers of users to simultaneously use satellite capacity. The advantage of DAMA is its efficient use of transponder resources, by assigning capacity only when a user has traffic to pass.

Network Hubs

An additional consideration is the topology of the network. Will the signals be routed through a central hub, or will all communication be point-to-point? Will a central switchboard need to be staffed, or can connections be automatic? Finally, can every user afford a hub; can costs be reduced?

One approach to solving all these challenges is the fully automatic hub, shared between many users and interconnected to the PSTN. An intelligent network master controller at the hub coordinates and controls all traffic. Vehicles can communicate to their stations by calling into the hub and then dialing into the PSTN. Costs are reduced between the users of the shared hub.

A New Network

GTE Spacenet, in conjunction with several major news organizations, has recently commissioned several voice networks operating on Transponder 1 of GSTAR II.

These networks are SCPC/DAMA full mesh networks. The full mesh capability of the network ensures that, while all calls are controlled and monitored by the hub station, any site may communicate with any other without need to "double-hop" through the hub.

If a vehicle communicates to the hub, it can elect to call into the PSTN. Conversely, any telephone in the PSTN can direct dial into the hub, and be connected to the vehicle. This operation is completely automatic, requiring no human intervention.

NETWORK DESCRIPTION

Figure 1 shows an overall view of the satellite news gathering network. Each network consists of three elements: satellite news gathering vehicles, the satellite, and the hub earth station.

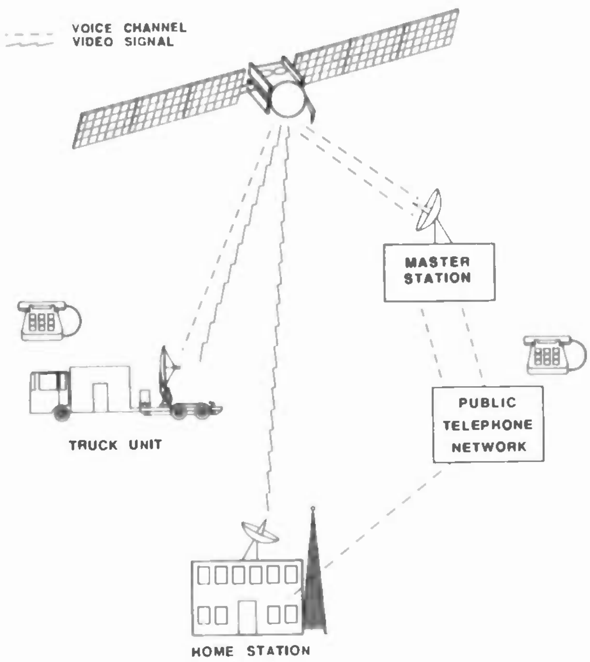

Figure 1
Satellite News Gathering Network

Hub Earth Station

The primary hub station for all networks is located in McLean, VA. As seen in Figure 2, the hub has several groups of baseband equipment, a radio frequency (RF)/antenna group, and a private automatic branch exchange (PABX).

The hub baseband equipment is a group of FM communications equipment. Each channel in the group consists of two units. A trunk control unit (TCU) contains all the telephony and voice frequency (VF) processing circuitry. The TCU interfaces directly with a two or four wire trunk coming from the PABX. The output of the TCU is full duplex, which is interfaced to the

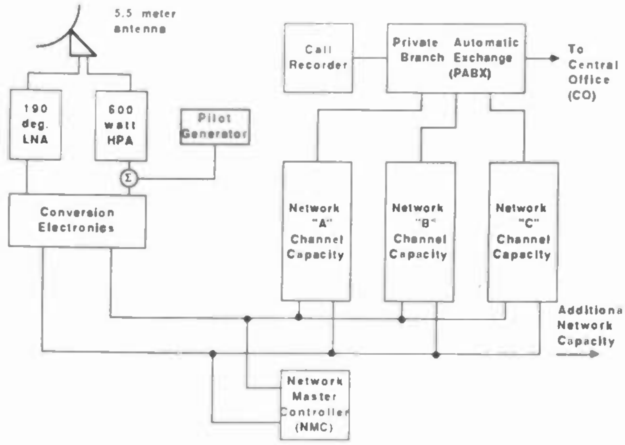

Figure 2
Hub Earth Station

voice channel unit (VCU). The VCU is essentially a frequency agile companded FM modem, which receives and transmits at a nominal 70 MHz intermediate frequency (IF).

A number of these VCU/TCU pairs are placed in a common rack, and serve as the hub circuit capacity for a specific network. Each VCU/TCU pair is programmed with the unique identifying number called the "site ID." The site ID is how the circuit capacity is identified to the network master controller (NMC).

Network Master Controller

The NMC controls the operation of all the networks. Located at the hub, the NMC consists of standard TCU, and two VCUs dedicated to its communication requirements. The master is responsible for:

a. monitoring and displaying status and roster information of all the circuits installed at a specific site ID

b. servicing every request to pass traffic

c. making frequency assignments for traffic

The master communicates to all sites via a outbound channel dedicated to it. Likewise, each site ID reports to the master on a polled basis on a common inbound channel. Each site is assigned a time to report to the master, and it is during this interval that the master receives information specific to the reporting site.

Public Automatic Branch Exchange (PABX)

All of the baseband equipment in each site ID at the hub has a corresponding trunk group assigned to it in the PABX. The PABX is used instead of direct assignment of central office trunks to ensure a working trunk can always be routed to a satellite channel requesting service. Additionally, when an satellite news gathering vehicle makes a call to the PSTN, call information is forwarded from the PABX to a call recorder.

RF Equipment

The output of all the VCUs in Mclean are combined, and upconverted to Ku-band. An additional Continuous Wave (CW) signal is inserted in this composite signal, which is called the pilot. The pilot serves as a frequency reference, by which all VCUs derive accurate frequency information.

The hub equipment is designed with full consideration of redundancy. The entire site is operated with uninterruptable power (UPS). Conversion electronics, high power amplifier (HPA), low noise amplifier (LNA), and system pilot are redundant. The 5.5 meter is antenna equipped with a full deice system.

Vehicle Equipment

Figure 3 shows a representative satellite news gathering vehicle with the voice communication package integrated. The principal pieces of the package are the baseband equipment, conversion electronics, and a 5W solid state power amplifier (SSPA).

Baseband equipment for the vehicle is available in up to four duplex channels. As in the master, each TCU/VCU is programmed with the site ID number assigned to that vehicle. This is how the vehicle or equivalent site ID is recognized by the master. Each TCU in the vehicle can be directly connected to a telephone instrument through an appropriate interface circuit. If additional telephone set capacity is required, a small PABX can be included in the implementation.

The output of the TCUs in the vehicle are upconverted to Ku-band, and amplified by a 5 watt nominal SSPA. This signal is then combined via a filter/diplexer with the video RF signal.

The filter/diplexer is essential, as compared to simple power combining or use of directional couplers, because of the minimal losses that must be incurred. Although the video HPA could also be used for the voice circuits, such use presents nontrivial complications in the area of the stable and repeatable uplink power levels, as well as backing off the video HPA to avoid intermodulation.

Network Operation

A call placed from a vehicle to the PSTN progresses as follows. The originator in the vehicle takes the telephone instrument "off-hook" and dials the hub destination site ID number. The off hook is translated as a burst of information to the NMC requesting service with the destination site. The master then informs the destination site of imminent traffic, and assigns a frequency pair for the traffic to pass. When destination goes off hook, the caller can access the PSTN, or coordinate a video feed. Upon call completion, the master will sense the instruments going back on hook, and take down the circuit.

For a party in the PSTN to access a vehicle requires a call to the hub. After issuing a series of access codes, the caller is connected to the first available circuit in their hub. From that point, the vehicle can be direct dialed by issuing the site ID, and extension, if any.

SPACE SEGMENT DESIGN

The voice circuit space segment design is based on integrated optimization of uplink and downlink noise and "spacelink" interference contributions. Spacelink interference is all interference mechanisms and consists of uplink and downlink adjacent-satellite interference, cross-polarized transponder interference, adjacent-channel interference, and uplink HPA and downlink spacecraft TWTA intermodulation interference. Given assumed characterizations of adjacent-satellite, adjacent-transponder, cross-polarized transponder, and adjacent-channel interference contributions, link impairment mechanisms may be reduced to three generic segments: 1) uplink noise, 2) downlink noise, and 3) intermodulation interference. These segments are interactively optimized in consideration of the design objectives and constraints summarized in Table 1.

Uplink Noise Considerations

The basic considerations that are factored into the uplink design are listed below:

a. uplink C/No performance (clear sky and faded)

b. maximum available vehicle and hub HPA power

c. vehicle and hub transmit chain feed losses

d. multiple carrier HPA output back-off requirements

e. edge-of-coverage uplink sensitivity.

The GSTAR series spacecraft are configured with a series of seven 3-dB ground-commandable attenuators. Through analysis, a determination was made that GSTAR II, Transponder 1 operation with a 9-dB pad would provide good uplink performance for both the inbound and outbound

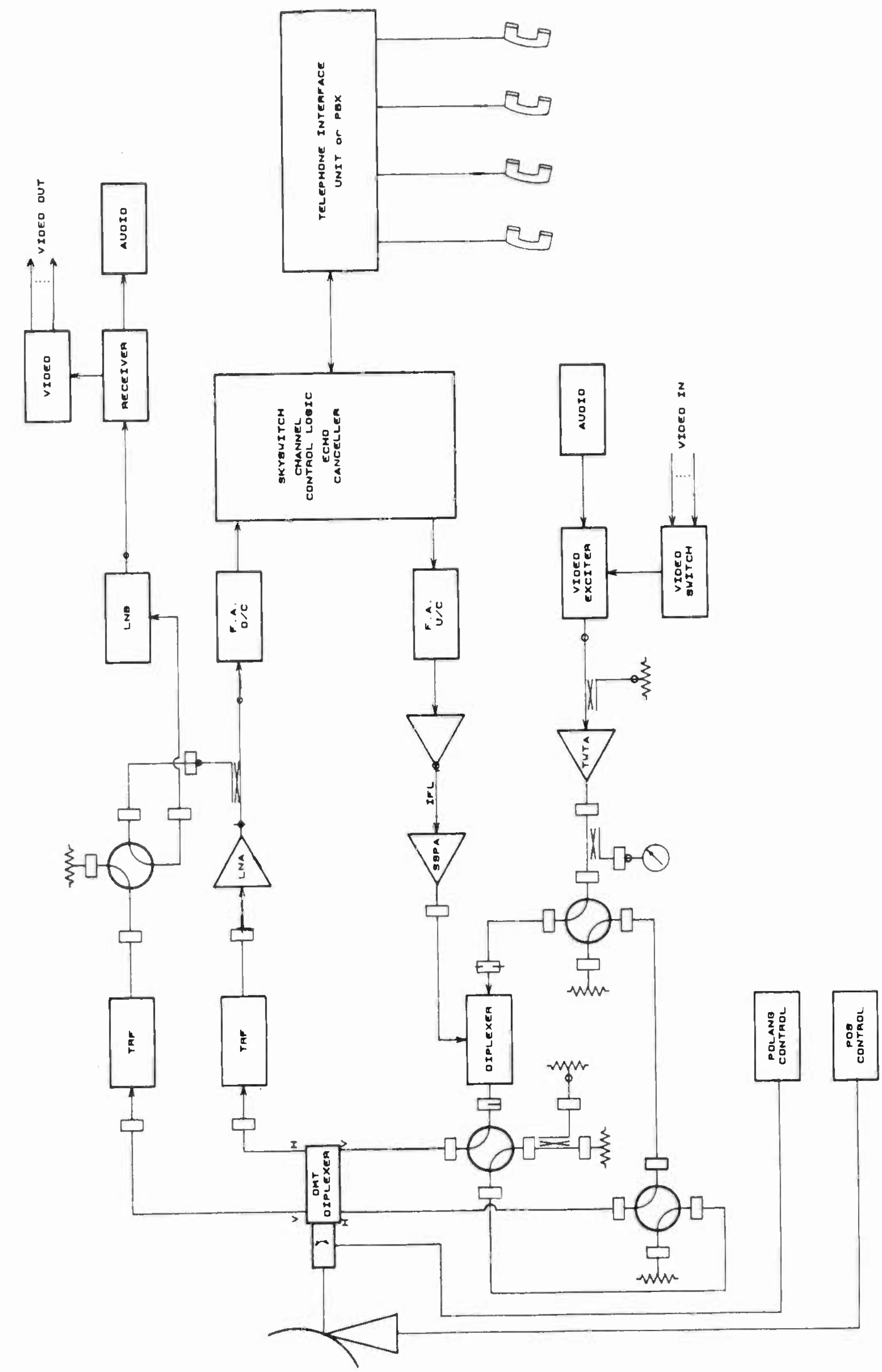

Figure 3
Vehicle Equipment

Table 1
Design Objectives/Constraints

Overall

S/N Voice = 50 dB (clear sky minimum)

Availability (half-link)* = 99.9% (excluding rain zones D3 and E)

Hub

Spacecraft G/T:	2.1
Spacecraft EIRP:	44.0
No. Duplex Circuits:	>350 (w/voice activation)
Available HPA Power:	600 watts
Antenna Diameter:	5.5 meters
Receive G/T:	29.7 dB/K (clear)

Vehicle

Spacecraft G/T:	-1.3 dB/K (edge-of-coverage)
Spacecraft EIRP:	39 dBW (edge-of-coverage)
Number of Circuits:	4
Available HPA Power:	5 watts (SSPA)
Antenna Diameter:	2.4 meters (minimum)
Receive G/T:	22.3 dB/K (minimum)

*"Half-link" considers only uplink or downlink fading between the earth station site and the spacecraft.

voice links, while meeting vehicle and hub HPA power/backoff constraints.

Downlink Noise Considerations

The basic considerations that were incorporated into downlink design are listed below:

a. downlink C/No performance (clear sky and faded)

b. vehicle and hub receive G/T (clear sky and faded)

c. edge-of-coverage satellite down-link EIRP.

Since the transponder is operated in a linear region (5 dB total output backoff), the downlink power levels are linearly related to the uplink levels and vice-versa, i.e., a one dB increase in uplink power will produce a one dB increase in downlink power. Based on the downlink parameters above along with uplink power level considerations, the downlink power was adjusted to satisfy the overall performance objectives.

Intermodulation Interference Considerations

The primary interference impairment within the system designer's control is intermodulation interference. Spacecraft TWTA operation with a total output power backoff of 5 dB satisfies transponder circuit capacity requirements while only producing worst-case carrier to intermodulation carrier-to-interference (C/I) levels on the order of 28 and 20 dB for the outbound and inbound carriers, respectively, assuming a Gaussian distribution.

Since the inbound carriers are the more sensitive of the two carrier types, intermodulation level analysis is conducted with respect to these levels. As noted above, the worst-case inbound C/I of 20 dB was based on an assumption of equal-spaced carriers producing a Gaussian distributed intermodulation noise distribution. With judicious frequency assignment derived from proven intermodulation software analyses, actual TWTA-produced intermodulation C/I levels greater than 23 dB can be achieved. Therefore, 20 dB can be employed as an overall intermodulation C/I objective for the more sensitive, inbound links, allocating 23 dB to spacecraft TWTA intermodulation products and 23 dB to uplink HPA (vehicle and hub) intermodulation products.

Furthermore, half of the uplink intermodulation interference contribution is allocated to the vehicle uplinks and half to the hub uplink HPA. Therefore, the inbound carrier uplink intermodulation C/I contribution from the hub HPA to the inbound carriers is specified to be 26 dB or greater, and the aggregate inbound carrier uplink intermodulation C/I from the vehicle uplink HPAs was specified to be 26 dB or greater.

Considering DAMA and voice activation uncertainties, a worst-case scenario of two intermodulation products falling within an inbound carrier frequency slot is assumed. This dictates that the worst-case intermodulation product produced by any single vehicle HPA produces an intermodulation C/I of 29 dB.

Based on the required uplink intermodulation constraint of 29 dB and an average uplink EIRP of approximately 37 dBW per carrier, an uplink EIRP intermodulation carrier level density constraint of 8 dB per 25-kHz is derived.

Summary Link Budget

A summary of pertinent link budget parameters is provided in Table 2. Table 2 was developed incorporating the values in Table 1 as well as the results of the above discussion.

VEHICLE CONSIDERATIONS

To ensure consistent and reliable voice communication from the vehicles, several key performance parameters have been identified. The following comments are applied to all vehicles that use the networks.

Table 2
Link Budget Summary

	Hub to Vehicle 5.5-2.4M	Vehicle to Hub 2.4-5.5M
Transmit E/S Antenna Diameter (m):	5.5	2.4
Receive E/S Diameter (m):	2.4	5.5
Highest Baseband Freq (kHz):	3.1	3.1
Peak Deviation (kHz):	9.2	9.2
Required C/N (dB):	6.5	6.5
Required C/No (dB-Hz):	50.5	50.5
Required S/N (dB):	49.7	49.7
Link Availability (%):	99.9	99.9
Saturation Flux Density (dBW/m^2)	-88.5	-85.1
Carrier Flux Density (dBW/m^2):	-121.1	-125.6
Saturated Downlink EIRP (dBW):	39.0	44.0
Output Backoff/Carrier (dB):	28.1	36.0
Carrier Downlink EIRP (dBW):	10.9	8.0
Uplink C/N Ratio (dB):	21.1	13.2
Downlink C/N (dB, clear):	11.5	16.1
Intermodulation C/I (dB):	28.1	20.1
Other C/I (dB):	24.9	21.1
Total Link C/N (dB, clear):	10.8	10.5
Total Link C/No (dB-Hz, clear sky):	54.8	54.5
Link S/N (dB, clear):	54.0	53.7
Uplink Margin (dB):	4.5	4.5
Downlink Margin (dB):	4.8	6.6

Antenna

One of the most important considerations when purchasing a satellite news gathering vehicle, is the satellite antenna and its mounting system. The antenna must comply with the Federal Communications Commission Rules and Regulations Part 25, Section 25.209 Antenna Performance Standards. This requirement must be met with the antenna in the final vehicle-mounted configuration as delivered to the end user. It is not sufficient that the antenna meets this requirement based on antenna manufacturers range test data if the antenna tested was some other configuration (i.e., tested antenna was 2-port feed and supplied antenna is 4-port feed with motorized polarization).

A key factor which affects the selection of antenna size is the expected geographical operating location of the vehicle. A vehicle that will be used near the edge of the satellite footprint or in a higher rain area should have a larger antenna with more gain.

Once the antenna size is set, the specific antenna/vehicle combination can be selected. The vehicle and the antenna mounting system should be designed to accurately and repeatedly deploy the antenna and keep it pointed at the satellite even during high wind conditions or personnel movement within the vehicle. Even the most compliant antenna, if not properly mounted, will not meet the cross polarization isolation or adjacent satellite isolation requirements. It is recommended that a structural analysis be performed on the final vehicle/antenna combination by a licensed civil engineer to guarantee compliance with antenna pointing accuracy requirements.

Receive RF Chain

In order for the voice communications package to operate satisfactorily with the Spacenet hub in McLean, Virginia, certain minimum performance specifications must be met.

The vehicle receive system figure of merit or G/T must be a minimum of 22.3 dB/K. This requirement can be met with most 2.4 meter antenna using a 160 degree Kelvin LNA if the insertion loss between the antenna feed and the LNA input is kept below .5 dB. Larger diameter antennas can be outfitted with higher noise temperature LNAs to maintain the same G/T or use the 160 degree K unit and benefit from increased downlink margin.

The voice communication system requires a nominal 12 GHz input into the high stability, low phase noise downconverter. If the video receive system require an L-band input, then a low noise block converter (LNB or LNC) is required in addition to the LNA. The 12 GHz output of the LNA can be split using a coaxial coupler with the through port feeding the voice communication system downconverter and the coupled port feeding the LNB. The coupler and LNB can be mounted either on the antenna or inside the vehicle depending on vehicle design and other requirements such as opposite polarization access switching.

Transmit RF Chain

The voice communication system should employ a separate SSPA to develop the necessary gain and uplink power. The major reasons for the separate unit are minimizing the third order intermodulation (IM) distortion and maximizing the uplink amplitude stability. This design allows for the greatest operational flexibility as well, by allowing video to be fed to either uplink polarization while the voice carriers are feeding the vertical polarization. Use of the SSPA enables use of the voice system soon after the antenna is peaked on the satellite.

The output of the 5-watt SSPA is fed to a transmit filter diplexer which combines the voice communications system SCPC signals with the video and audio signals when the video feed is on the same polarization. The design of the filter diplexer allows low loss combining of the voice carriers in the 50 MHz band centered on 14030 MHz (GSTAR III, Transponder 1) with video carrier(s) in the band 14125 to 14500 MHz (GSTAR II, Transponders 3 through 8). The key here is low loss combining. Because of the limited output power available from the SSPA when it is operated at backoff sufficient to keep the intermodulation distortion levels below acceptable limits, only 1.6 dB total loss from the SSPA output port to the feed input of a typical 2.5 meter antenna can be tolerated. Allowing .6 dB loss for the filter diplexer leaves only 1 dB for additional waveguide and switching losses. Of course if a larger diameter antenna is used, the allowable loss increases dB for dB as the uplink gain increases. This allowable uplink loss will govern the mounting location of the 5-watt SSPA and the filter diplexer. With the smaller diameter antennas (2.4 meter) both units should be mounted right at the antenna either on the feed arm structure of one antenna design or on the backside of the antenna reflector of other designs.

The filter diplexer incorporates a 10-watt load with an over-temperature sensor/switch that opens when the load reaches 80 degrees C. This switch must be wired into either the video TWTA RF inhibit circuit or the high voltage shutdown circuit. This arrangement provides reverse power protection for the 5-watt SSPA output by shutting down the video TWTA output if the video exciter is accidentally tuned to a frequency which is passed by the filter. If this were to happen, the power would pass through the filter in the reverse direction toward the SSPA.

Transmission Line Loss

The allowable losses for the transmit and receive interfacility links (IFLs) from the upconverter output to the SSPA input and from the LNA output to the downconverter input can be a maximum of 10 dB with a typical 2.4 meter antenna. Using .25 inch diameter heliax cable would allow maximum lengths of between 20 and 25 feet depending on other losses from couplers, switches, etc. Larger antennas can tolerate slightly higher IFL losses.

Figure 3 depicts a typical 4-port feed vehicle configuration. This configuration allows the video TWTA to be switched to the vertical feed port either via the filter diplexer or by bypassing it. This capability along with RF patching or switching between the upconverter output and the TWTA input, allows the voice carriers to be amplified by the TWTA in case of a failure of the SSPA. This would also allow the voice signals to be fed to a transponder out-of-band from the filter diplexer. This switching allows the simultaneous uplinking of the voice signals on vertical polarization with the video signals on either polarization. A high power dummy load is included for use by either the TWTA while retaining full voice communications capability or by both amplifiers when combined through the filter diplexer.

On the receive path, one LNA is used to provide the amplified 12 GHz signals for the voice system downconverter from the horizontal feed port. A single LNB which is fed from either feed port provides the signals for the video monitoring receiver.

Many other configurations are possible, allowing almost unlimited flexibility and redundancy. Dual LNBs could be installed, for instance, to allow simultaneous video reception from both polarization or dual TWTAs in either a redundant or phase-combined configuration could be provided to increase transmit system availability. Two (2) port antenna feed systems can also be used with a reduction flexibility and ability to access the video transponders on the other polarization.

CONCLUSION

The GTE Spacenet GSTAR II satellite news gathering networks are a significant advance in satisfying the voice and data communications requirements of mobile news gathering vehicles. Vehicle equipment guidelines have been presented, which promote consistent and reliable performance for news gathering operations.

ACKNOWLEDGEMENTS

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PRE-FABRICATED STUDIOS FOR RADIO BROADCASTERS: HOW AND WHEN TO USE THEM EFFECTIVELY

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ABSTRACT

Several years ago the author began experimenting with a variation of prefabricated industrial sound proof isolation booths for radio studios. This paper discusses how this technology responds to the pressures of modern studio placement, offers practical advice on using the devices effectively, compares prefab performance with traditional built on-site studios and compares the cost of both methods.

HOW PREFABS MEET THE PRESSURES OF CONTEMPORARY STUDIO CONSTRUCTION

The art of building excellent studios for radio or TV audio use is well defined. Beginning with the heavy cloth draping of the 1920's we have evolved a mature technology based on proven principles of mass, isolation and absorption to achieve exceptionally good and highly predictable results in studio construction. The principles involved are well understood by a large number of consultants, architects and engineers. We know how to build excellent studios. Unfortunately a number of pressures are being brought to bear on the studio construction process that make it difficult to continue practicing some of the methods that we know work. In seeking ways to cope with these pressures we began experimenting with the use of prefabricated controlled audio environments originally used for industrial sound control. While the technology is not perfect we have found that this alternative significantly expands our arsenal for dealing with the expanded pressures of this industry in the 1980's while maintaining a liveable performance level. This paper will discuss how this technology responds to some specific contemporary pressures and offers some practical experience-based suggestions on utilizing this tool in facilities construction.

The first group of pressures this alternative can help answer can be grouped broadly under the heading of expanded non-use-specific location complications. At one time radio stations were located in buildings built specifically for them. As our industry has evolved, this has changed. Greater equipment reliability, improved remote control systems and high quality studio transmitter link technology have freed radio from many of its early location restrictions. Our staff sizes have also shrunk significantly. We now have the freedom to locate our operating centers almost anywhere. Add to this the pressure from sales and programming to locate the station close to the clients and in the nicest possible surroundings and we find ourselves wanting to place studios in generic buildings that are built to accommodate basic office space or seeking to creatively re-use existing areas.

The new locations typically carry restrictions that make traditional studio construction techniques difficult. These include relatively light floor loading, limited space between floor and ceiling decks, unsophisticated community air conditioning and heating systems and lease restrictions forcing tenants to return space to the landlord in pre-renovated condition at the end of a lease term. Prefab technology has helped us respond to some of these concerns.

Floor and structural loading in many buildings are not sufficient to support the floating concrete slab, multiple sheet rock walls, double framing, and lead sheathing that are included routinely in traditional studio construction. The structures are sized for desks, partition walls, people and small to medium office equipment. While the prefab studios are not light on average, they present only about forty percent of the load of a traditional studio. This loading per square foot places the prefabs in roughly the same area as a densely populated, highly mechanized office. This is much closer to the capabilities of most modern structures. Those that are deficient can usually be brought up to this level with moderate expense.

Deck to deck clearance is another concern. Traditional construction practices eat up space. The floating floor takes eight to ten inches and the anchors for the floating ceiling take two to three feet. Space must also be allowed for the large volume, low velocity air ducts and sound traps in the HVAC system. Prefab techniques help reduce this requirement. The sandwich floor plus the support rails and isolators take about six inches. The ceiling of the prefabs are self supporting so there is no requirement for hangers. The ducting for the supply and return air is incorporated into the structure ceiling panels reducing the clearances needed for elbows. If you use the silencer boxes supplied by the vendor they are built into the ceiling panels of the studio holding the height needed above the studio to under twelve inches. This makes it practical to place a studio in a building with only twelve foot deck-to-deck clearance.

The built in ducting and silencer help in another way. If you ask the house mechanical engineer with most buildings to design an HVAC system that meets the strict isolation and system noise requirements of a broadcast studio it will create a problem. This work is highly specialized. If the house firm has the expertise you are certain to pay a substantial fee. You are more likely to have to call in an outside firm. This further increases the cost because they must become familiar with the building and their work will be subject to review and approval by the house firm upon completion. This will be an additional cost to you. By comparison the prefab is easy. The manufacturer specifies the quantity, temperature and static pressure of the air to be supplied. The building engineer can deal comfortably in these terms. This will achieve more satisfactory performance and greater consistency in results.

The positive effect of using prefabs on the landlord tenant relationship are somewhat subtle but definitely real. To a landlord a perfect tenant quietly occupies a space for the term of a lease while requiring a minimum of support services, attracting as little attention as possible and returning the space in as close to original condition as possible. If your plans include pouring new concrete floors, floating ceiling from hangers and creating ten inch thick lead lined walls you can expect most building rental agents to get very unfriendly. The basic nature of prefabs seem to reassure building managers. The fact that they can be disassembled and removed much like office partitions creates the visions of unencumbered clear-walled spaces that populate facilities managers dreams. A lot of objections, concerns and performance bonds seem to magically disappear.

During any project there is constant pressure to cut the amount of time taken to complete. Traditional studio construction techniques require great care on the part of the construction trades. This care takes time. Prefabs help eliminate some of this pressure because they go up fast. It is not unusual for a crew to complete a two or three studio complex in less than two weeks. This simplifies scheduling the other trades for their work in the studios because there is much less dead time while the studio structures are being completed. The studio portion of the project shifts from being a construction item to an installation by a single party. This makes it much easier to manage in the flow of the project.

There is another closely related consideration. As mentioned above, the traditional studio construction requires a great deal of care. Crafts people must pay strict attention to construction specifications that are complicated, time consuming and unfamiliar. Without close on-site expert supervision a lot of seemingly trivial (from a construction worker's point of view) but crucial items will be ignored. This supervision takes time and costs money. Without it the quality of the results will vary widely from project to project depending on the construction team involved. Modular studios do require care in their assembly but there are advantages. The assembly is usually done by a factory trained team augmented with craftspeople from the local hiring hall. The team is familiar with the problems and pitfalls of the prefabs and are better prepared to deal with them. A second advantage is the level of expertise required. Since much of the precision work affecting the integrity of the prefabs is done at the factory, the work performed on site is much closer to standard construction practice than the traditional studios. You are far less likely to suffer from depending on the varying quality of labor from contractor to contractor and one part of the country to another. We can be far more certain of the results we can achieve.

The question of predictability extends beyond the construction crafts. While we are all familiar with the basic principles of sound control, it is a complex and difficult art. When we are supervising the design of a facility we are faced with a dilemma. Most architects do not have the expertise to handle the complex acoustical problems of even the simplest studio. Outside consultants are expensive, hard to locate and difficult to evaluate. They often have trouble conveying all of the necessary concerns to the architect. It is not unusual for a number of sets of drawings to pass back and forth between the parties before the problems are all addressed. Coordinating this flow takes more of the project team's time and paying the per hour charge for the consultant and the architect eats at the budget. If short comings are discovered after construction, there tends to be a lot of finger pointing as the various parties try to cover their own positions. This delays achieving final solutions. The prefabs simplify the process. The project team can give the architect a simple laundry list of support requirements from the prefab's spec sheet. The performance of the studios rest solely on the prefab manufacturer and supplier. While they are not simple catalogue items, the design process is greatly simplified and the predictability of results improves.

The advantages of prefabs continue after the initial construction is completed. In traditional construction, future maintainers of a facility must rely on "as-builts" to plan for maintenance or alterations. Unfortunately collecting the job-site red lines and compiling good working drawings is tedious work that pays little reward to those who must do it. It is also an area that can be cut easily when a project is running over budget or beyond deadline. A future facility manager may have precious little information about exactly how a given part of a studio was actually built. Since the prefabs are factory made there is much less chance for on-site variations. Even if there are no drawings at all, a future planner can use catalogue sheets and be almost certain how a studio is built.

PRACTICAL CAUTIONS IN USING PREFABS

While prefab studios may be an attractive alternative for some projects there are some cautions to observe. These include maintaining adequate isolation, being more aware of the internal reflections in the studio, keeping an eye on sizing, watching equipment loading and maintaining adequate floor stability.

Isolation is the primary principle used to control sound transfer in prefab studios. It is most important that the installation be planned to maximize this isolation. The studios are designed to be floating rooms. To live in a radio station they have to be tied into the overall building. Doing this without compromising the studio isolation requires some care. First, it is nice to be able to mount the studios in a pit. By planning the pit height correctly the studio floor will match the corridor floor height exactly. This eliminates the clumsy step up into the studios and permits a sliding plate tied to the studio to bridge the door gap. Use carpet on the floor by the outer studio walls. Run the carpet up to the gap and cut if off. Attach the vinyl base to the outer studio wall allowing it to just graze the top of the carpet nap. The interface between the studio and the ceiling in the surrounding area is also crucial. Don't allow the ceiling contractor to attach supporting track to the outside studio walls. The ceiling should come within about one half inch of the studio outer walls and stop. Where the studios approach a wall or each other, isolation can be maintained by using a gasketed steel plate. The plate is attached to either the studio or the structural wall at one end. The other end is gasketed and floats against the wall. Hard conduit is another route for sound to follow. All connections to the studios should be made using flexible conduit.

Minimizing penetrations is another important concern in maximizing prefab performance. A normal studio can tolerate a fair amount of conduit runs and other breaches of its integrity because the mass of the walls is great and the sound from minor leaks tends to be damped out close to the source. The lighter prefabs are more sensitive to noise that gets through the barrier. There are several tricks to minimize the problem. Make only a single penetration for AC power. Use surface mount fixtures and wire mold on the walls and ceilings. Floor outlets to feed the equipment can be run using flat conductor cable systems under the carpeting. Audio and control wiring can enter the room through conduit or Walker style cable duct under the floor. The best route for either conduit or duct is through the floor to minimize studio to studio transfer through breached sidewalls.

Don't lose sight of the basic laws of sound reflections. When one can stack panels like blocks to build a studio one can easily forget to practice the common sense rules of studio design. The rooms should not be square to avoid standing reflections. The panels are available in both a solid steel and perforated with fiberglass surface. They should be arranged to reflect and absorb as needed for whatever character of room resonance is desired. Since the windows are flat instead of slanted one must be careful to place a soft surface opposite the window or the classic slapshot echo will occur. Attention should be paid to speaker placement and wall treatment to insure that the sound from the monitors is accurate.

Particular care must be taken in the installation of any vibration producing items in the studio. Since the box is isolated the goal is to attach as much as possible directly to the studio walls and ceilings. You do not want to do something like tying the monitor speaker mounts to a structural ceiling through the studio panels. This will transfer sound past the studio walls where it is more likely to travel into an adjacent space. The same caution applies to the other machines that produce high levels of vibration. Placing these items carefully can contribute significantly to reducing sound transfer.

Floor loading is another touchy area in prefabs. We are all used to sketching out furniture layouts for traditional studios so that isolation risers of sufficient strength can be installed. Prefabs are no different. If there aren't enough isolation rails and acoustic isolators in a particular part of the studio it will tilt. Since the structure is relatively light the entire studio will flex. This causes everything from sound leaks due to separating panels to jammed doors and cracked windows.

Care must also be paid to the finish of the floor supporting the studios. The acoustic isolation/support rail system is designed to work on a perfectly flat support slab. Any deviation in surface finish will require shimming during installation. This increases costs and increases the chances of failure due to shifting.

Careful shimming and leveling of the floor of the prefabs is important for another reason. As we have noted several times before the prefabs achieve their sound proofing by isolation instead of mass. The steel sandwich floor is far more flexible than a suspended slab. This can cause a serious problem with some types of turntable mounts. The best solution is a high mass, high vibration isolation mount installed away from heavy traffic areas of the room. Adding extra isolation mounts in the immediate area of the TT cabinet also helps to stabilize the platform.

There is one other consideration to keep in mind while planning your studio layout. The standard prefabs studio panels are 4' x 8'. Any size panel can be made to complete a studio wall of any length but if you have the luxury of designing your own space it is far cheaper to hold to standard dimensions so custom panels are unnecessary.

While you are juggling the studio sizing you might also give some thought to decoration. The prefab studio is ugly. Spending some time, effort and money to make it more pleasant will pay dividends later. Several items help. First, there is no reason not to sheetrock over the metal panels on the studio exterior. This covers the clumsy elevated seams on the panels so the studio walls match other walls in the complex. On the inside a nice natural oak chair rail and baseboard can add considerable warmth to the studio. The studio walls can be painted any color. A bright accent wall and careful selection of a base color can make the studio a more pleasant place to work. Painting the studio interior requires additional caution. Be certain the painters thin their paint before applying it to the perforated panels. Thick paint will clog the holes and acoustic absorption will be reduced substantially.

Be prepared to mediate a jurisdictional dispute if the studio construction project is union manned. The assembly of prefab studios does not fall squarely under any construction union jurisdiction. Normally carpenters, sheet metal workers and sheetrockers will try to claim the work. Occasionally iron workers will also try for a claim. There is no easy solution. The traveling crews are perfectly capable of doing all the work. The question is how much local talent will you be forced to add and from what union. Reaching an agreement up front can save unnecessary on-the-job confrontation and help insure that the assembly is done with care.

PERFORMANCE OF PREFABS VERSUS TRADITIONAL ON-SITE BUILTS

No matter what advantages factory built studios offer for the project planner, they would be worthless if their performance was sub-standard. This has not proven to be a problem in the projects where we've used prefabs. The design target we adopted when using traditional construction techniques was to obtain a noise rating of NC 20 with all machines in the studio operating, one person in the room, the HVAC operating in cooling mode and the monitor in the adjacent studio playing off-air (heavily processed) audio at full jock level (105 db or so). Our usual results measured between NC 24 and NC 28. The same design goals were applied to the prefab studio projects. We were surprised to find that our average results in the prefabs was an NC 21 to an NC 25. The difference seemed to be in the nature of the noise. The on-site studios seemed to suffer most from reflection of in-studio sound and from air conditioning background noise. The prefabs seemed to perform better in these areas but have more difficulty handling low frequency leakage from adjacent studios. In spite of these observations the performance of both types of studios proved to be entirely satisfactory in service.

COST COMPARISON

Comparing construction costs is frustrating because there are so many variables. Identical projects in Denver and Cleveland may differ in cost by as much as 30% due to local conditions. The prefabs make budgeting easier because the price is the same no matter where they are built. To decide on future action we spent a good deal of time trying to quantify the cost differences between prefab and regular studios. After weighing things as carefully as we could we finally concluded that the cost difference between prefab and regular construction favors the prefab by about 8%. In the the overall budgets of most construction projects, this would not have enough impact to be a major factor. The decision to go routine construction or prefab should therefore be made on the basis of suitability to application not cost or performance.

DESIGNING NEW YORK'S LARGEST AM/FM STUDIO FACILITY

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As the nation's telecommunications systems expand, so do the complexities which broadcasters must accommodate to remain competitive. The technical concepts and features which have been incorporated into the new WNYC AM and FM facilities represent, on a large scale, considerations which will have application in the future design of stations finding it necessary to reach beyond their record libraries and telephones for their programming.

The design program for WNYC required the flexibility of a radio network to originate and redistribute programming not only by AM and FM transmissions, but also by terrestrial and satellite links. Designed by Northeastern Communications Concepts, Inc. (NCC) for super-station capability, WNYC's facilities place no restrictions on either programming formats, technical quality, or simplicity of operation.

HISTORICAL PERSPECTIVE

In June of 1922, the New York City Board of Estimate appropriated $50,000 for the purchase and installation of equipment for a City owned and operated broadcasting station. Two years later, after battling AT&T's patent rights which then controlled equipment use and the air waves, and after changing its call sign from CONY, WNYC was granted further permission from the Board to purchase broadcasting equipment without engaging in standard bidding procedures. A similar provision was enacted by the Board 60 years later, to assure that the quality standards set by NCC's designs could be met.

WNYC began broadcasting at 570 kHz on July 8, 1924. Amidst its pioneering efforts in classical music entertainment, the station's programming often fell victim to the needs of the City's police and fire departments, not to mention those of its prominent politicians. When Fiorello LaGuardia ran for mayor in 1933, his platform included plans for the sale of WNYC. But by the time he ran for reelection in 1937, he became its staunchest proponent, mandating that the station "serve the people" as a City instrument, in the arts, social services and the celebration of good government and citizenship.

Encouraged by the popularity of its informative and classical music programs in the New York area, WNYC began high quality broadcasting on the pre-war FM band on September 1, 1943, and on television channel 31 in 1962. Despite changing its AM frequency to 810 and later to 830 kHz, and its FM frequency to 93.9 mHz, relocating its transmission sites, and occasionally upgrading its equipment through 1952, the WNYC studios remain today where they opened in 1924, on the penthouse 25th floor of the landmark status Municipal Building, located at the foot of the Brooklyn Bridge in Manhattan.

When WNYC's earth station began operations in 1980 establishing it as one of 15 uplink stations in the National Public Radio Network, and later as a founding member of the American Public Radio Network, the technical demands of the satellite age were suddenly thrust upon one of the most antiquated broadcast facilities in New York.

The aging studios which had been constructed as a WPA project during the Great Depression, and the Master Control Room equipped by Gates in 1952 were still in operation through 1985. Left channel stereo FM was mixed through a vacuum tube Gates console, while the right channel was routed through a Shure M67 mixer. The mineral wool acoustical material in the studio walls had sunk to the bottom of the cavities in a useless and compacted mass. Ad hoc editing and listening booths were set up wherever there was space for man and tape machine. While it is a tribute to the operations staff that the WNYC's sonic quality belied the sad state of its studios, it was obvious that the situation could not continue indefinitely.

On the rebound from New York City's fiscal crisis during the 1970's, Mayor Koch approved the City's first major capital investment in the station in thirty years. At last the facility which served the world's largest city well for so long, and which bore a striking resemblance to a Flash Gordon movie set, could be retired.

Supplemented by revenues from the newly formed WNYC foundation and public grants, the long term goal was to create a non- commercial broadcasting entity which is sustaining, and completely free from reliance on the City's operating and capital budgets. As part of this project, A.W. D'Alessio Associates (now NCC Inc.) was retained as broadcast consultant to design and manage the renovation of the 1952 facilities

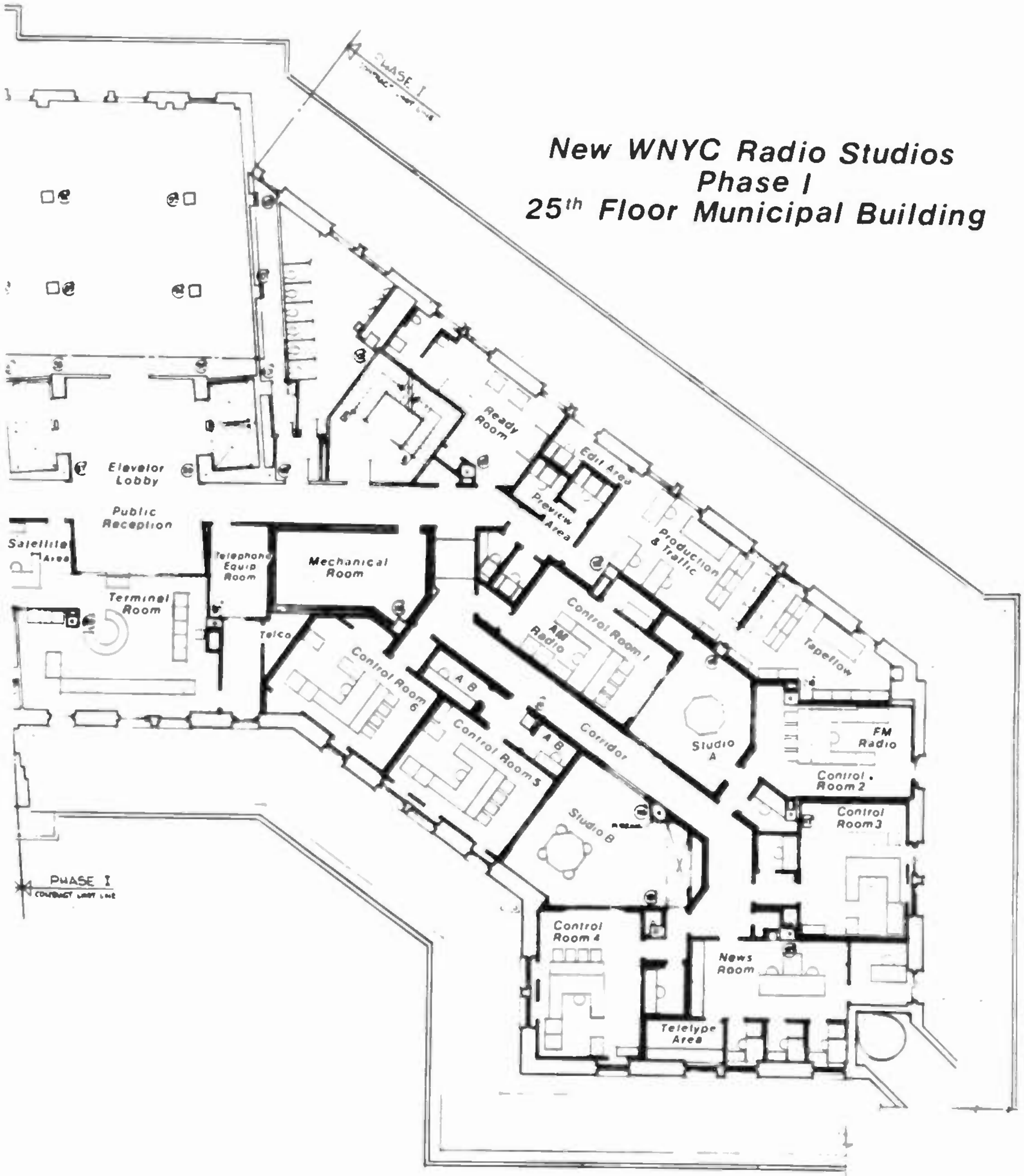

Figure 1. WNYC Operations Area Space Plan

and to increase the facility's technical potential and quality, while reducing the manpower needed to staff the operation.

Classical music has been predominant on WNYC-FM's programming schedule for years. In order to attract discriminating listeners, wide dynamic range and frequency response with the lowest possible levels of distortion are absolute imperatives. Of paramount importance for the new studio plan was the requirement for attaining the highest possible standard of audio quality. Since this consideration included reproduction from recorded media in addition to live broadcasts; significant emphasis was placed not only on high performance equipment and systems design, but also on maintaining stringent acoustical characteristics.

As a source of City news and information itself, and as a distribution point for commercial programming circulated across the nation by several satellite carriers, WNYC forms the hub of a telecommunications network connecting virtually every other New York City radio station with nation- wide program services. Therefore, the quality and reliability of the new WNYC would not only be evident in local broadcasts, but also in satellite program syndication and distribution.

RENOVATION vs. INNOVATION

When NCC was retained by WNYC, its renovation project had already fallen almost twelve months behind schedule. The architectural program requiring six spacious combo control rooms, two talk studios, a live performance studio, and a host of support facilities had been established by the operations department; however previous attempts by several consultants to develop a schematic design consistent with WNYC's requirements had been scrapped by the station management.

Without the advantage of selecting a site more suitable for studio construction, NCC faced the challenge of redesigning the facility in a space originally intended for small public offices. The site was an awkward shape, congested with columns, and with floor to roof heights of less than ten feet in certain areas. A water tank on the roof served the entire building through massive plumbing which would have to be relocated away from the studios. Space for the large air conditioning ducts and silencers required to keep air noise at a minimum was extremely limited. The building had achieved landmark status, which placed additional restrictions on the design, and it was located directly above a noisy subway station.

Project Phasing

After careful review of the finances and the total space of over 23,000 square feet available for the project, NCC recommended that the renovation be divided into two phases. Phase I would establish the basic operations facilities of the new plant within 10,400 sq. ft. of space then occupied by a mix of radio and television offices and edit areas on the 25th floor, and relocate the radio management to new offices on the 26th floor in the Municipal Building Tower. Phase II would demolish approximately 6,400 sq. ft. in the area of the old radio studios on the 25th floor after the cut over to the new facilities; and would establish new offices for the technical personnel, a new record library and maintenance facility, and the large live performance studio and 24 track control room. The design and construction of the Phase I studio complex is the focal point of this paper.

Within two weeks NCC developed an architectural schematic responsive to WNYC's program requirements and acceptable to its management. It resulted in the the Phase I facility depicted by Figure 1. When Mayor Ed Koch dedicated the Phase I studios on December 11, 1985 in memory of Fiorello LaGuardia, WNYC was established as New York's largest AM/FM broadcast facility, even before Phase II is funded and built.

WNYC Space Plan

Visitors and guests stepping off the elevators on the 25th Floor are greeted by the futuristic but tasteful decor of the public Reception Area, which architecturally states the philosophy behind the renovation. Accenting this statement is a 20 foot glass vision panel bearing the WNYC logo, which permits an expansive view of the Terminal Room. Together with the adjoining Satellite Uplink Area, Telephone Central Office Equipment Room and Internal Telephone Equipment Room, the Terminal Room serves as the technical nerve center for the entire operation. Guests may either proceed to the studios or relax in the Ready Room lounge until air time.

The high-tech atmosphere is limited to the Reception Area, Terminal Room, and corridors. All other facilities are intended to convey a feeling of warmth, with pastel colors accented by oak wood trim, for the comfort of those who must spend protracted hours in the operations area.

The main corridor offers access to combo Control Room/Studios 1 through 6, each of which features ac-

Figure 2. Studio B as viewed from Control Room 4. Portions of Control Room and Announce Booth 5, the Main Corridor, Studio A and Control Room 1 can be seen through the vision panels.

commodations for four guests in addition to a host/engineer. While programming can originate from any of the six control rooms, Control Rooms 1 and 2 are reserved primarily for combo operations for live FM and AM programming, and either can originate broadcasts from Studio A, which is located between them. Similarly, Control Rooms 4 and 5 share Studio B, which is acoustically large enough to accommodate a six foot grand piano and a small chamber music ensemble. Studio B is shown in Figure 2.

At the end of the corridor is the news complex, consisting of the News Room, News Director's Office, three News Edit Booths, and a Wire Room for teletypes and storage. The entire operations area, including the Terminal Room and News Room feature raised computer type access flooring, under which run the audio and control lines for the entire plant.

Just off the entrance to the main corridor is the Production and Traffic complex, which includes five Edit and Previewing Booths, an open office area, and the Tapeflow Area, for interim storage of program tapes and tape syndication.

EQUIPMENT & FURNISHINGS

The combined requirements of simplicity, quality, and flexibility, dictated the design of custom or customized equipment, when certain features and functions could not be procured from the usual pool of broadcast products.

Control Rooms

Consoles. The two On-Air control rooms and two production control rooms are equipped with 22 input BMX Series III consoles, which were added to the BMX console line by Pacific Recorders & Engineering Corp. as a result of the specifications developed by NCC for WNYC and in response to technical requirements for broadcast consoles established at the BBC. The resultant consoles feature expanded bus architecture to handle multiple mix-minus feeds for telephone talk programs, superior control logic, monitor facilities for both a studio and announce booth, improved communications paths, a stereo solo bus, and two effects buses. By incorporating these additional features into the original BMX line, one console type serves efficiently as both an on-air and production board.

For more advanced production requiring multi-track tape recording, two Pacific Recorders ABX consoles were selected by WNYC for Control Rooms 4 and 6. The ABX consoles provide eight channel multi-track mix buses and built-in accommodations for equalization and signal processing modules. As an air console, the basic functions of the BMX consoles are retained in the AMX series, to provide technical consistency for the operators.

Microphone On/Off/Cough switches and headphone level controls may be operated remotely from any control room, announce booth, or studio talent position. Each studio and announce booth is also provided with remote control of the console timer logic, talkback and monitoring functions from a small desk top mounted control turret.

Approximately 400 long frame I/O patch points for all peripheral equipment and console signal paths are provided in each control room to increase flexibility, to provide trouble shooting convenience, and to quickly alleviate any emergency conditions which may arise.

Tape Machines. On-air and production control rooms have two or three Studer A- 80 reel-to-reel stereo tape recorders respectively. Control Rooms 4 and 6 have each been designed to accept an 8 track 1" recorder. Each recorder is wired to a Fabec-Gotham noise reduction card frame, which is nominally outfitted with a Dolby A type, Cat #22 card. The NR frames can also accept DBX K-9, and Dolby Spectral Recording cards in addition to any of the other noise reduction systems compatible with this format. Flexibility in changing noise reduction formats is essential to WNYC for compatibility among the classical music affiliates who communicate their programming through WNYC's facilities.

Three Pacific Recorders stereo Tomcat cartridge playback decks provide the choice of 7.5 or 15 ips, L/R or matrix operation in all control rooms. Control Rooms 3 through 6 each also feature one record deck. Rack mounted, Tascam 122B cassette recorders with +4 dBu balanced inputs and outputs, as well as mounting and electronic provisions for future R or S format digital tape machines were also specified for each control room.

Microphones and Speakers. Extensive subjective testing was employed in choosing the microphones and loudspeakers for the studios and control rooms. While no microphone is ideal for all applications, AKG 414EB large diaphragm RF condenser microphones were chosen as the best compromise to all around performance for speech and music. The ability to tailor their polar and low frequency response without changing capsules, in addition to their moderate price and subjective performance outweighed the minor inconvenience of needing a wind screen for explosive guests.

Unlike other microphones of similar design, AKG does not provide a low frequency rolloff at 30 Hz. For this reason we do not recommend them for studios where low frequency acoustical noise is not tightly controlled. Tailoring the response at either of the microphone's two selectable frequencies of 75 or 150 Hz with a 12 dB per octave slope is too severe for critical voice applications, when the only consideration is removing the sub-sonic room tones that cause speaker cone and VU meter wobble. All microphones throughout the plant are mounted on spring tensioned extension arms bolted to the control room, announce booth and studio cabinetry.

As with microphones, loudspeakers fall prey to subjective preferences, particularly in view of the fact that laboratory tests of them do not correlate well with their perceived characteristics. Aggravating this problem, consider that most loudspeaker colorations change significantly among rooms of different sizes and acoustical loading properties. However, substantial agreement among the participants in the loudspeaker listening tests resulted in selecting Studer BR series loudspeakers as the most natural reproducers for the loudspeaker placement and mounting techniques, and studio and control room acoustic environments created for WNYC. The speaker cabinets were specified to be suspended from the ceilings on articulated yoke mounts, acoustically isolated from the structure by combination spring and neoprene devices.

Peripherals. The casework housing the consoles and peripheral control room equipment was custom designed with plastic laminate finishes and oak splashbacks and trim, color coordinated with the pastel colors and wood finishes unique to the individual control rooms. The modular cabinetry was assembled on-site into a monolithic work station, integrating specific areas for the operator, interview guests and equipment. It consists of a four position interview table opposite the console and operator, and two equipment returns.

One equipment return extends from the floor to a foot above the 29" work surface to provide four EIA mounting bays for the cartridge and cassette decks, compact disc players, the jack fields, distribution amplifiers, distribution switcher terminals, expansion space, and audio processing equipment including stereo limiters and graphic equalizers. The face of the return below the counter height slopes at a 12 degree angle from vertical for the operator's convenience.

The return opposite the sloping cabinet houses two Technics SP-15 turntables, which may be concealed by a hinged cover in the production control rooms, to protect their delicate mechanisms when not in use, and to provide the additional work surfaces necessary for certain production activities. The turntables are mounted at counter top level on 70 pound mass loaded platforms, independently and resiliently attached to the cabinetry to reduce sensitivity to shock and vibration. WNYC engineers have replaced the original sand ballast with compacted lead, to increase the air circulation beneath the turntable drives. Two small but accessible rack bays below the turntables house the turntable preamps, noise reduction frames, and other peripheral equipment which need infrequent attention.

Two pedestals flanking the operator's knee space below the console contain the monitor, cue, and communications power amplifiers, in addition to the console power supplies.

Figure 3. **View of Terminal Room from monitoring console. The reception room vision panel and WNYC logo are at the left. Mounted on the wall between the tape machine and equipment racks are the HVAC controls.**

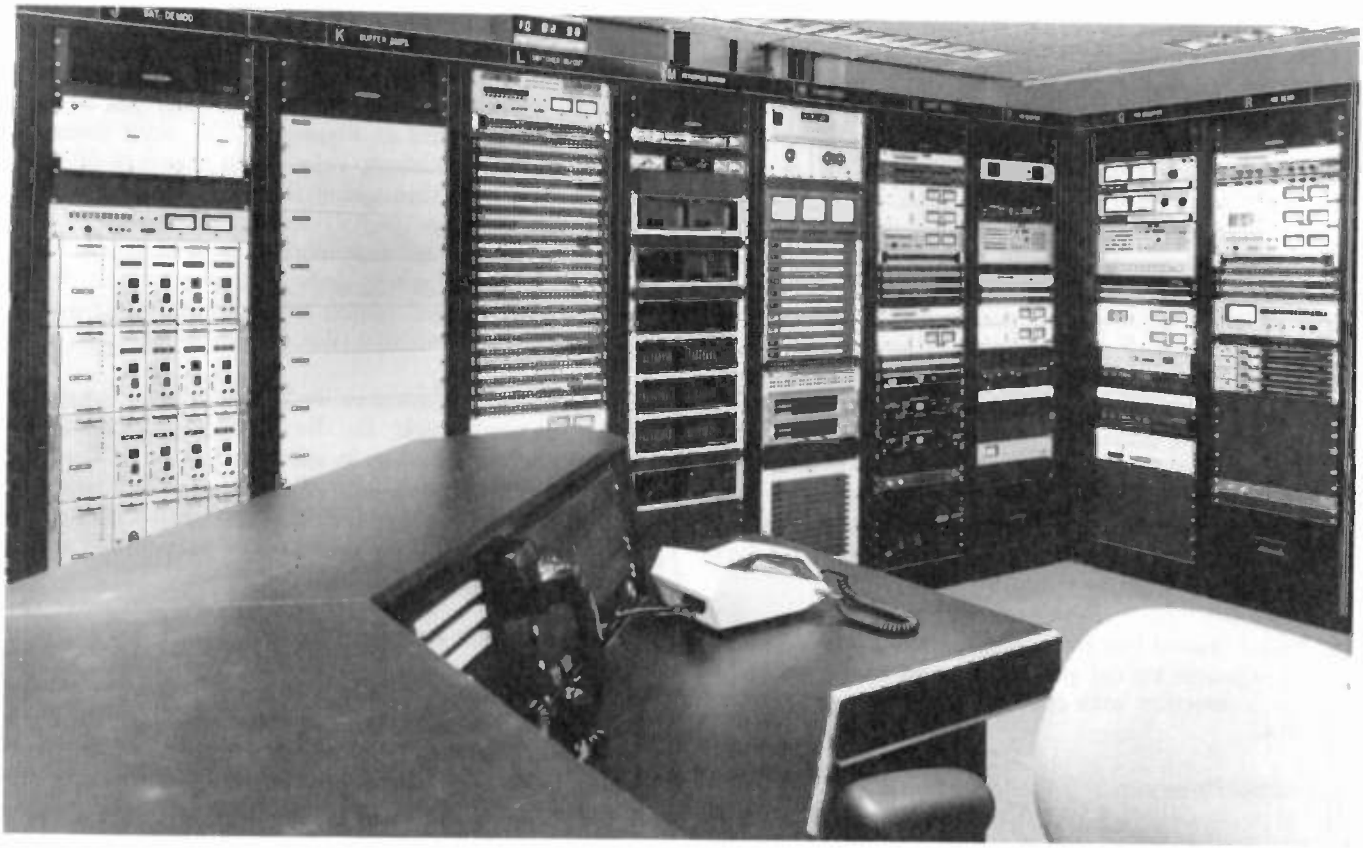

Figure 4. **View of Terminal Room looking toward the trasmitter control racks. The fourth rack from corner houses the distribution switcher matrix, flanked by its jackfield access and the video switcher.**

All together, the control room cabinetry provides the equipment and personnel accommodations listed above, including 115-1/2 inches of accessible EIA rack mounting space within a compact floor area of less than 90 square feet.

Studios

Studios A and B each feature an octagonal interview desk. Generous space is allocated around these tables for a host and up to seven guests. The size and shape of the table was designed to maximize acoustical separation between its positions minimizing phase cancellations among the microphones.

The studios are employed primarily for broadcasts or recordings requiring seating capacity beyond that of the control rooms, or for live music. Since each studio is shared by two control rooms, a custom changeover switcher transfers the microphone lines, monitor and communications signals, and all remote control and tally lines to the appropriate control room console. The specifications required that the microphone numbers of the table positions also change with the transfer, so that they read correctly from left to right from the engineer's view point in either of the two opposing control rooms.

Figure 5. **News Edit Booth.**

The transfer switcher, monitor amplifiers and logic power supplies are located in a small rack in the corner of each studio. The desk and equipment rack casework is identical to that of the control room furnishings, with the exception of the colors.

Edit Facilities

In order to reduce the work load in the four fully equipped production control rooms, two edit suite areas were provided. The three news room edit booths were carefully equipped to accommodate the requirements of producing news actualities and phoners. As shown in Figure 5, each features an MCI reel-to-reel tape recorder, a Tomcat cartridge recorder, a cassette recorder, distribution switcher access, a timer, and a six input news mixer. The custom cabinetry provides additional space for cart racks, monitor speakers and amplifiers, telephone recording equipment, writing space, a typewriter return, and storage space for tape and editing supplies.

An additional five preview and timing booths are located adjacent to the production and traffic areas for use in connection with continuity and tape distribution control.

Terminal Room

Visitors viewing the Terminal Room from the entrance lobby often mistake it for a master control room, which connotes memories of radio days past when a full time engineering presence was required to manage the operation from a central area. Instead, WNYC's Terminal Room is just what the name implies, the area of the technical plant where all of the audio and control lines terminate for internal or external distribution. It is from this room that up to 200 telephone program lines, 30 satellite links on Westar IV, audio lines serving 7 control rooms and remote control of the AM transmission site in Greenpoint, Brooklyn and the FM transmitter at the World Trade center are managed.

The partial view of the Terminal Room, shown in Figure 3, illustrates the view from its monitoring and communications console. The five equipment racks facing the console contain eight Revox PR-99 reel-to reel tape recorders, used primarily for recording programming downlinked from the satellite. The machines may be operated by either local or remote control from an NPR data channel carried on the satellite. Two slow speed PR-99 decks are used to log various WNYC programming and certain City government proceedings. Each pair of machines is fed from a Distribution Switcher Terminal (DST), with their outputs routed back to the switcher for distribution within or outside of the facility.

The next three racks to the right of the tape decks provide interface to and from the program lines in the adjacent Telephone Equipment Room. Satellite demodulators are located in the next two adjacent racks, keeping all incoming audio signal paths within one area of the Terminal Room.

The buffer amplifiers, patching, and monitoring associated with the main distribution switcher can be seen in the view depicted in Figure 4. For news coverage and television simulcasts, video from a variety of sources is distributed throughout the plant by a 3M 15x12 Audio-follow-Video (AFV) switching system. The remaining racks in the corner contain redundant (primary and emergency) audio processing, noise reduction, monitoring, and remote control equipment for the AM and FM transmission sites.

Technical conveniences in the Terminal Room include an inter-rack tie line system, local stereo loudspeaker monitoring in most racks, and video and critical listening monitoring of the entire plant from the central console. Also included are pull out storage drawers and AC power convenience receptacles accessible from the front of many of the racks.

Satellite Uplink Area

Adjacent to the Terminal Room is a seventh control room dedicated to satellite uplink operations. While it performs functions that are an extension of the Terminal Room operations, it was decided to keep this area separated by glass doors since uplinking some types of programming may require careful quality control monitoring and a degree of production.

Although a four meter satellite downlink dish is installed on the roof of the Municipal Building, frequency congestion in Manhattan mandated locating the uplink in Brooklyn. Remote controls for STL microwave link to Brooklyn are located in three equipment racks in the Satellite Area.

The room is also outfitted with a BMX III console, outboard equalizers and audio processing, audio and video DST's, and four A-80 tape machines for feeding programs to the Satellite.

DISTRIBUTION SWITCHER

Design Criteria

Superior management of the comprehensive range of audio distribution requirements was a crucial consideration during the design of the WNYC systems. After an exhaustive market search for a large switcher capable of handling the projected signal load, it was concluded that the only functionally adequate switchers consisted of not much more than audio circuitry stripped from an Audio-follow-Video design, and could not meet NCC requirements for size, audio quality, flexibility, cost, and simplicity of operation simultaneously.

Since WNYC was designed to accommodate digital audio equipment as part of its future, it would have been a travesty to contaminate the technology with noise and distortion from the system distributing the signals from digital devices in the analog domain.

Additionally, the signal traffic through WNYC is a mixture of stereophonic and monophonic signals. Requirements for keeping stereophonic lines paired without limiting monophonic capabilities are diametrically opposed. Stereo sources should be so presented at the remote left and right output points, while mono sources should be presented on both the left and right output destinations simultaneously. It was considered a compromise to flexibility to dedicate a portion of the switcher to stereo operation and another part of it to mono operation, since the proportion between the two modes was not anticipated to remain constant with increasing stereo demands. Utilization of certain satellite links and telephone lines complicated matters further, since they may be employed as stereo pairs for some services, and independent mono lines for others.

Cross point conservation also entered as an economic factor. Monophonic switchers with the same number of channel inputs as a stereophonic switcher require twice the number of cross points. Simple arithmetic serves as an illustration: for example, a mono 192 input, 64 output matrix requires 12,188 cross points (192 x 64 = 12,188). But by switching the same number of inputs and outputs as stereo pairs, only 6,144 cross points are required (96 x 32 x 2 = 6,144). Doubling up mono sources to both the left and right inputs of a stereo switcher wastes cross points. Large matrix arrays are not only expensive, but can also contribute to the degradation of cross talk and signal-to-noise characteristics of the switcher.

For achieving uniform outputs, input levels such as those from telephone company program lines require gain adjustment. Therefore, a further requirement for providing 30 dB of gain was necessary to raise the lowest acceptable telephone line levels to the plant standard of +4 dBu. (This standard was chosen as the best compromise between noise and headroom considerations, particularly for loading outgoing telephone circuits.)

Above all else, all of the equipment specified for WNYC had to be "user friendly", since the bulk of the operation is handled by semi-technical operations personnel, not engineers. Switcher status had to be evident and understandable in plain English, and the number of controls to operate it had to be kept at a minimum.

In order to achieve these criteria, specifications for a custom audio distribution switcher were developed by NCC. The resultant switcher is based on a cascaded matrix with 6,144 cross points, which operates under the control of a 16 bit microprocessor. The I/O architecture operates in a mono/stereo hybrid mode, switching 192 inputs to 32 remote points either monophonically, as stereo pairs, or mono L+R sums. Specially modified DA-150 buffer amplifiers supply up to 30 dB of gain in two switchable ranges to maintain signal-to-noise performance.

The matrices and microprocessor were developed and built by NTP Electronik A/S in Copenhagen, and the remote Distribution Switcher Terminals (DST's) and buffer amplifiers were developed and built by Pacific Recorders & Engineering. Figure 4 illustrates the compactness of the switching system power supplies, CPU, and matrix, all of which occupy only 42" of vertical rack space. The quality of the switching system is typified by the most difficult parameters to achieve: cross talk separation in excess of 100 dB through 15 kHz and a dynamic range of 115 dB, which is 22 dB greater than 16 bit digital audio!

Until the introduction of the new International Tapetronics / 3M audio switcher introduced at the NAB Convention this year, no other manufacturer adequately addressed the difficulties inherent in distributing stereophonic and monophonic signals with a single switcher.

Operation

Operation of the switcher is performed at any one of the DST remote locations. Figure 6 shows six DST's in the terminal room. They drive six specially modified Pacific Recorders SDA-8 distribution amplifiers, each one of which is capable of driving 16 telephone lines. Selector keys on each DST are marked with the names of ten groups of sources: EXTERNAL, SATELLITE, HOUSE PAIR, UPLINK, TELEPHONE, TAPE, PROGRAM, NETWORK, AM and FM, in addition to

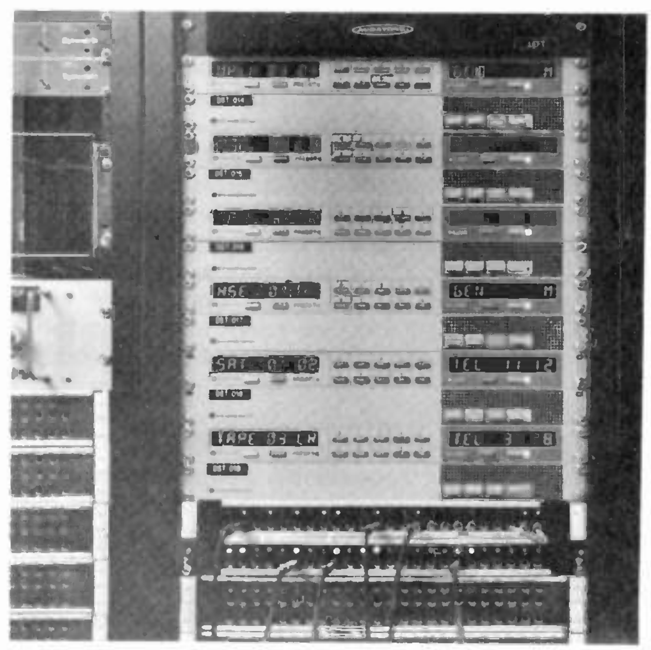

Figure 6. Six Distribution Switcher Terminals (DST's) in operation in the Terminal Room. Tie line annunciator LED's can be seen directly below them.

being labeled with the numerals 0 through 9. Depressing one of these keys prompts the operator for a line number. For example, depressing the PROGRAM key followed a "0" and a "6" presets the DST to switch to program audio from Control Room 6. The preset display also prompts the operator that the Control Room 6 source is normally stereo, but provides him with the option to select only the left, or right channels, or sum them to mono, by activating the appropriate DST mode key. Depressing the TAKE button switches the preset display to the line display, and the switcher output at that DST location to the preset source.

Two DST's are located in the primary AM and FM control rooms, and one each in control rooms 1 through 6, and the three news edit booths. Other locations for DST's are at the ten dubbing recorders and at other racks in the Terminal Room and Satellite Uplink Area.

Although the switcher is provided with redundant power supplies and other protections, in case of catastrophic failure, two passive networks of 24 tie lines each are available. They link the various racks in the Terminal Room with the Satellite Room and all control rooms. A custom annunciator network driving an LED above each jack indicates whether that particular tie line is in use anywhere in the plant. As a result, unintentional mults and cross patches are easy to avoid when selecting a tie line.

ACOUSTICAL DESIGN

Keeping It Quiet

Consistent with WNYC's outlook for the future incorporation of digital audio technology and the wide dynamic range it implies, an acoustical design was developed by NCC for maintaining residual noise levels below a criterion of NC-20. Limited ceiling height precluded the use of fully floating "room within a room" studio construction. This presented a challenge not only to isolating the studios from structure borne vibration, but also limited the effectiveness of the partitions to isolate noise between adjacent areas. Since each studio is shared by two control rooms, sound transmission from competing operations would have to be limited by innovative partition design to make up for inability to isolate them on independent floating floors.

Additionally, the studios would be unavoidably sandwiched between two sources of vibration. The only location for the air conditioning equipment was on the building roof, the floor directly above the studio complex. The operation of the IRT subway 25 floors directly beneath the studio complex may have presented additional acoustical difficulties, had it not been for the massive terra cotta arch construction of the Municipal Building, typical of high rise construction in 1911 when it was completed.

NCC therefore developed an acoustical design with resiliently suspended gypsum ceilings to isolate any noise or vibration generated by the HVAC equipment on the roof. Special acoustical hangers were designed by NCC to suspend the isolation ceiling from the terra cotta arch above, which in turn supported an acoustical finish ceiling.

Massive dry wall partitions with large air cavities damped with internal fiber glass insulation, and resiliently attached to the structural floor would provide the necessary sound transmission loss between rooms, at the expense of requiring an inordinate wall thickness and mass.

As a result, no studio or control room ceiling or wall is mechanically coupled to those of any adjacent area, despite the lack of a floating floor. Partition construction consists of multiple wythes of 1" thick gypsum core boards and 5/8" gypsum panels, supported by heavy gauge steel studs. Complete with the acoustical finish materials discussed below, the vertical mass of the construction between the studio and control room areas is approximately 300 pounds per lineal foot. The floor loading this represents could not be supported by most modern high rise office buildings.

Doors And Windows

The use of acoustically damped sound locks for access to all acoustically sensitive areas permitted the use of attractive wood doors carrying a sound transmission class rating as low as STC-36.

As is evident from Figure 2, the extensive use of architectural glass creates a feeling of openness, which departs from the visual isolation typical of many studio installations. It is a major factor in reducing staff fatigue by presenting a pleasant work environment. However, the use of expansive vision panels necessitated creating the asymmetrical shapes of Studios A and B to avoid developing flutter echoes between parallel window walls. The panels are double glazed with custom manufactured, 7/8" and 1" thick, composite laminated glass, to match the sound transmission characteristic of the partitions in which they are mounted.

Room Tuning

When planning acoustical spaces with volumes less than 5,000 cubic feet particular attention must be given for handling low frequency reverberation. The tendency of the average size radio studio or control room to unevenly reinforce certain frequencies below 200 hz can create a boominess which is further exaggerated by multiple microphone pickups. As opposed to music recording, the spoken word should be transmitted to its destination without coloration (unless it is desirable as a special effect). For these reasons, acoustic wall treatments were designed by NCC not only to keep speech frequency reverberation time low, but to also provide a high degree of low frequency absorption.

After considering the acoustic impact of the windows, floor, ceiling, and equipment in each space, the walls were treated accordingly. The familiar looking, cloth covered wall treatment above chair rail height in the studios and control rooms is actually composed of varying thicknesses and densities of fiberglass materials with a flush surface for visual consistency. It provides a significant amount of absorption from very high frequencies to as low as 80 Hz. Below chair rail level, resonant panel absorbers covering damped cavities of varying sizes provide additional low frequency absorption, and a durable surface in an area that is subject to physical damage.

Although difficult to calculate or measure in small rooms, the overall reverberation time (T60) in the studios and control rooms is in the order of 210 msec. Although this produces a relatively "dead" acoustical characteristic, it encourages inexperienced or timid radio guests to project their speech to desirable levels, since they do not hear their own speech reinforced by the room acoustic. A low T60 also extends the distance between the speaker and his microphone before an "off-mike" sound is evident, and minimizes the transmission of noise caused by tape machines, cooling fans, and other equipment located close to the microphones. The wall treatments also dispose of unwanted early reflections which can cause comb filtering and deep coloration of the loudspeaker sound. While favorable in most instances, the low reverberation time requires that loudspeaker monitors be driven by higher than normal size amplifiers, in order to attain the same perceived levels of operation that would occur in a room with a higher T60.

CONSTRUCTION

The efficacy of the best acoustical designs can be compromised by poor workmanship and techniques during construction. Unfortunately, the City bidding procedures required awarding the construction contract to the lowest bidders, which precluded contracting firms with studio construction experience. Continuous delays ensued, prohibiting construction progress beyond a few weeks work. To no avail, the City used its best efforts to build what had been designed.

Many of the difficulties were attributable to the New York State Wicks Law, which requires all public construction projects in excess of $50,000 to be awarded to separate general construction, electrical, mechanical, and plumbing contractors. The usual Prime/Sub-contractor arrangement is prohibited, with a City designated resident engineer assuming the managerial role of the prime contractor. Mayor Koch has consistently railed against the Wicks law as the greatest impairment to public works projects. The complexities of the WNYC project were no exception, and beyond the experience of the resident engineer to coordinate.

As a remedy, NCC was retained by the City to perform coordination, management, and regular inspections of the studio construction for quality and progress. This intensive effort was matched only by the magnitude of paper work and red tape required to satisfy the government bureaucracy which existed outside of WNYC. While the pace of construction never caught up to that more typical of the private sector, the construction phase was substantially completed in time for the delivery and installation of the broadcast equipment.

EQUIPMENT INSTALLATION

Fearing that a repeat of the construction dilemmas would occur with the procurement and installation of the broadcast equipment and systems, WNYC sought and was granted permission by the New York City Board of Estimate, to issue a proprietary bid for the bulk of the broadcast installation. As recommended by NCC, the bid was awarded to Pacific Recorders & Engineering on merit rather than government procedure. Pacific subsequently pre-wired and factory tested the control room sub-assemblies within the cabinet modules built by them. The WNYC Chief Engineer then assumed responsibility for the coordination of the joint efforts of the WNYC technical staff and Pacific Recorders to carry out the on-site installation.

The installation was successfully and substantially completed by the date set for the grand opening, with the last phase of the project being rescued by the same Board of Estimate action taken 60 years earlier to found WNYC.

ACKNOWLEDGEMENTS

I thank Charles E. Corcoran, WNYC Chief of Operations, for entrusting us for the design of new WNYC.

I extend my gratitude to Sidney Feldman, former WNYC Chief Engineer. Through his dedication to WNYC, and extensive dialog with us, we were provided with a strong appreciation for the station's technical operation and objectives, upon which we predicated our design.

I am also appreciative of the efforts of Andrej Tobiasz, Department of General Services Resident Architect, for conscientiously applying the skills of his trade and government experience during the construction of WNYC.

Finally, without the enterprise of WNYC Chief Engineer Ernie Dachel during the final design and installation of the broadcast equipment, our work would have been left undone.

THE NEW PBS MULTICHANNEL TELEVISION TECHNICAL FACILITIES

Alfred A. Norcott and Gregory T. Tarver
Public Broadcasting Service
Alexandria, Virginia

In October 1984, following a fire in the upper floors of the building it occupied, smoke and water damage forced the Public Broadcasting Service to vacate its administrative offices and technical operations facilities. A decision was made shortly thereafter to relocate PBS to new offices in Alexandria, VA for the long term.

This paper describes the planning, design, and construction of the new physical plant and technical facilities for PBS. Significant economic and technical goals were set and achieved and an aggressive schedule has been met. A review of the project goals and highlights, facility description, and commentary on the results are presented herein.

OCTOBER 15, 1984

On the night of October 15, 1984 a multiple alarm fire burned through the upper floors of the U.S. Postal Service building at L'Enfant Plaza in Washington, D.C. The building had no sprinkler system. As a result, the fire gained a substantial foothold before effective suppression could be applied. Hundreds of thousands of gallons of water were poured in by aerial ladder and by firemen who climbed the building's ten stories with hoses and oxygen masks. Although there were no serious injuries, PBS headquarters offices on the fifth and sixth floors of the building were severely damaged by smoke and water.

At the time of the fire PBS technical operations personnel were feeding the evening's time-zone-delay programming to the satellite network, from the technical center located on the second garage level underground. Torn between maintaining service and escaping the imminent hazards of toxic gases, electrical shock and the fire itself, they maintained programming to within ten minutes of the end of the night's schedule. When it appeared there was no choice but to abandon the facility they removed as much as could be taken of the precious library of current programming videotapes. By the following morning the building was unoccupiable and in-rushing water and smoke had rendered the technical center equipment useless.

Temporary operations began almost immediately with help from other public television facilities. Within four days a videotape truck loaned by ABC and a rented mobile unit were providing the majority of PBS feeds from the organization's uplink facility in Springfield, VA. A temporary technical center was installed in the tiny building and the staff dug in as the arduous recovery process began.

In early 1985 PBS began to search for new headquarters. The fire created an early opportunity to act on long-term space requirements: the fact that occupancy had been denied as a result of the fire, that the Postal Service was not going to renew PBS' lease when it expired in 1991, and that a long-term, more economically favorable lease could be negotiated elsewhere all contributed to the decision. Several sites in and out of the Washington area were investigated. Existing or soon to be completed construction was required, as the time available to withdraw from the existing lease was limited. After considerable investigation a new office complex in Alexandria, VA was selected.

Although the selected site was very desirable as standard office construction, it presented many challenges in the design of the technical facilities. Because of its recent experience at the L'Enfant Plaza location PBS had no desire to again locate its technical facilities below grade where cascading water could find its way. As a result, it was decided that the technical facilities would be located above ground, albeit in space that had originally been planned for typical office activities. The third floor was selected as the most suitable on the basis of available space and access. The columns on twenty foot spacing, the ten foot overall ceiling height, and window glass that completely surrounded the floor came with the package.

THE RECONSTRUCTION PROJECT COMMITTEE

In March 1985, the Reconstruction Project Committee was formed to act as the oversight organization for the design and construction of the new PBS technical center, and as a resource to PBS administration for the planning of all technical aspects of the relocation. Representatives from PBS Engineering, Technical Operations and Technical Maintenance staffs were included on the committee. The committee's primary role was to ensure that the design of the television facility was guided to the greatest extent possible by input from user organizations, so that the resultant system would meet all functional requirements, provide reasonable capacity for growth, and require minimal changes after installation. Although the "design by committee" process took longer on the front end, -- diverse interests represented on the committee sometimes inspired spirited debate over design issues -- the end product is something that all parties made a significant contribution to, feel satisfied with and is functionally appropriate.

Design Criteria

As noted, the technical facilities were located in what was originally intended to be standard office space. This alone would have made construction difficult, because of space limitations and the degree of completion of the building. The design requirements that the electrical and HVAC systems provide the highest degree of reliability and redundancy, and adherence to today's stringent building code specifications for technical occupancy construction and safety added to the difficulty.

The previous facility, completed in the late 70's, was very advanced for its time, but incorporated largely custom designed hardware and systems. The new Television Technical Center was required to perform all of the functions of the previous facility with ample room for growth and enhancement, but with the goal of employing as much standard off-the-shelf hardware as possible. This goal was met. The next phase of the project contemplates introduction of some of the very latest product and software developments to provide even higher levels of program distribution quality with increased efficiency.

BRADDOCK PLACE

Located at the Braddock Road Washington Metro Subway station just south of National Airport, PBS' new home is convenient to both public and private transportation. Completion of the building and build-out of PBS office space proceeded while the more complex third floor design was prepared. The third floor physical plant plans were completed in December 1985. PBS offices were moved to the site in March 1986, simultaneous with the start of construction on the third floor. PBS, with the cooperation of OMNI Construction of Washington, was able to independently procure the major HVAC and electrical equipment, avoiding substantial markups and saving over two hundred thousand dollars on the project. This practice is generally frowned upon in the construction business, but OMNI took a public spirited attitude toward this particular project, for which we are grateful.

The PBS technical facilities located on the third floor of Braddock Place consist of: the Television Technical Center, including the Technical Operations area, Equipment Center, Videotape Editing Suites, Teleconferencing Studio, and Technical Maintenance and Videotape Library support areas; the Computer Services data processing center, telephone switch room, Engineering Development Laboratory, and Technical Operations offices. Support areas off the floor include an electrical distribution/UPS room on the first garage level, standby generator and TVRO Satellite Dish at ground level, HVAC equipment at the penthouse level, and microwave relay tower on the main roof. The technical floor occupies approximately 21,000 square feet.

Electric Distribution

There are three tiers of electrical service for the technical floor: uninterruptible power, conditioned power and utility power. The main technical service feeder, at 480 volts, is separate from service to the rest of the building, coming directly from the complex's stepdown transformers. The 480 volt service is significant in that it is used to feed power from the garage level to the third floor where it is transformed as required for distribution to the load centers. This substantially reduced the cost of cable and installation and ohmic losses through reduced current. Redundant public utility feeders to the site were considered, but determined early on to be far too costly in comparison to providing standby power protection on site. The technical center and computer room are protected by a 160Kw Liebert uninterruptible power supply (UPS), which operates as a battery charger-inverter combination. If utility power fails, even for an instant, a battery bank in parallel with the rectifier-charger that is continuously in circuit maintains power to the output inverter. Because the system is fully regulated the inverter AC output never changes despite the interruption resulting from the loss of input. If the input power is lost for more than 15 seconds, automatic sensing circuitry in an 800 amp transfer switch will start the emergency generator. When the generator stabilizes to 60Hz, usually in a matter of a few seconds, the transfer switch connects the generator to the UPS, which then provides power once again from its input. The PBS battery bank will provide the full 160Kw output for five minutes, longer if the load is less. The generator also provides power for critical air conditioning and lighting. As the original pre-fire PBS generator had very low hours it was extracted from its indoor environment, reconditioned and encased for outdoor service, converted to 480 volts output and reinstalled at Braddock Place.

The second tier of electric service power is conditioned power for the editing areas, teleconferencing studio, and Engineering Development Laboratory. The system used is a 100 KVA Liebert Datawave, which will correct all electrical anomalies short of an interruption but including the loss of a phase.

The third, unconditioned tier feeds noncritical areas of the floor, mechanical equipment, and utility outlets for janitorial service and has ample capacity for future growth.

The technical center power distribution incorporates an isolated ground system. All television equipment and cable shields are connected to this ground and to no other. This ground system is wired in a multiple hub and spoke configuration and brought directly to the building service entrance ground.

HVAC

To provide precise temperature and humidity control, and operate the HVAC system in the most efficient manner, it was necessary to minimize the effect of solar gain and outside temperature as transferred through the windows that surround the third floor. Even although bronzed thermopane glazing was used throughout, without modification of the window construction, low outside temperatures would cause moisture to condense out of the 50 per cent relative humidity environment desired. To solve the problem, the perimeter of the entire technical area was given a second insulated partition with a vapor barrier. Existing window blinds were left in a closed position to maintain the outside appearance of the building. This construction also satisfied a requirement for the Halon fire suppression system in that the new outside walls provide an effective barrier against the escape of the Halon. All interior partitioning defining air conditioning and Halon protection zones were given similar insulation and sealing. The air conditioning system uses the same techniques as many modern computer facilities. The system supports PBS' 2000 square foot computer center, the videotape library, and the telephone switch room in addition to the television facility. Each technical area has its own redundant air conditioning units, which control both temperature and humidity precisely. Total system capacity is 170 tons. Heat is rejected from the conditioned area using a pumped water and ethylene glycol solution, which is cooled to outside air temperature on the roof. In the cooler months, the temperature of the solution is low enough for the system to remove facility equipment heat without HVAC unit compressor operation. This ability reduces operating costs dramatically. The units, Liebert System Three 20 ton models, provide conditioned supply air under the raised floor, and draw return air down through their top openings, from the room and the ceiling plenum. With the closed system construction and the use of a raised floor, the environment qualifies under the National Electrical Code for the use of non-plenum-rated cable in the underfloor area. Standard video and audio cable is used for all distribution run under the floor.

Fire Suppression

The original construction of the third floor provided sprinkler system protection only. The risk of damage and interruption of operations through accidental triggering of the system was determined to be too great to employ the sprinklers in the technical spaces. Although Halon was initially far more expensive than the sprinkler modifications that would have been required, its use was considered preferable in the final analysis because a discharge causes no damage of its own, and operation can be resumed immediately if there has been no serious fire. Local jurisdiction code requirements include maintenance of a reserve supply of Halon so that protection could be reestablished immediately following a discharge. Protection was also required both above and below the raised floor areas. The Halon detection and alarm system operates in two stages: the first detection sounds the alarm, notifies the fire department, and shuts down HVAC equipment. A second detection in the same area begins a 60 second countdown after which the Halon is released and all electrical power in the protected area is shut down. This two zone design is very effective in preventing unintentional release of the expensive Halon. Electrical power shutdown emergency pushbuttons are also required by code at all exits as protection against shock hazards in raised floor areas.

Interconnection

The PBS technical center is connected to the Springfield, VA uplink site, seven miles to the southwest via common carrier microwave. The original L'Enfant Plaza site used relay points at Tenley Circle (Washington D.C.'s in-town antenna farm) and an apartment complex in Bailey's Crossroads, VA. A direct line of sight to the uplink was not available. The new Alexandria location can see both of the original relay points with only a modest microwave antenna tower on the roof.

The microwave relay operates with four channels going to the uplink and three returning through the Virginia relay; two channels going and one returning via the Tenley site. These paths provide some diversity as it is necessary to use 11 Ghz frequencies on the uplink side of the relays, because of interference from the uplink to the preferred 6 Ghz frequencies used on the other paths. One advantage of the third floor location of the technical facilities is that the microwave equipment can be located on the floor instead of on the roof. Path loss was not significantly affected by the waveguide run required, and the expense of constructing a separate shelter for the microwave equipment at the roof level was avoided.

A direct shot to the uplink appears possible with additional tower height and this prospect is being investigated. For the long term, however, PBS believes that a fiber optic interconnection will provide the most reliable, transparent transmission. A study is underway to develop costs for rights-of-way and construction.

TELEVISION TECHNICAL CENTER

Having to relocate to Alexandria so unexpectedly presented an unforeseen opportunity to re-examine the technical services that PBS provides to its member stations and to redesign an aging facility with state-of-the-art technology. Among the considerations were the distribution and monitoring of increased hours of stereo programming and the distribution of additional self-supporting services. The choice of "major system components" (routing switcher, machine control system, master control, and automation) was made with the need for enhanced service capabilities in mind.

The following describes the criteria used in the overall design and the selection of major system components, the capabilities of the facility, and several of the most notable details.

Services Provided To Users

The primary service provided by the PBS Technical Center is the distribution of the National Program Service (NPS) to the more than 300 Member Stations. The NPS comprises all of the children's, cultural, news, and public affairs programming that uniquely define PBS. The program distribution function is analogous to a local station master control function. Programs on 1" tape are played back through the distribution system, monitored and a single feed is assembled through a master control switcher. The need to distribute to four time zones in addition to the original feed creates the need for multiple master control operations.

In the past, this multiple channel operation was accomplished through a pair of custom four channel automated switchers. A single operator would typically switch and monitor two main output channels while a second operator in the other identically equipped, redundant control room switched and monitored the other two main output channels. The remaining two channels in each of the switchers would ordinarily back up the main channels of the other control room.

Since the design and installation of the original plant, PBS began to distribute programs with stereo audio over the Digital Audio for TElevision (DATE) system. When stereo distribution was added to that plant, it required the addition of a separate stereo switching matrix to feed the DATE encoders. Today's routing switchers are all equipped for multiple audio levels.

The question of stereo monitoring resulted in a more far-reaching change in operating philosophy. It was agreed that as greater percentages of the schedule included stereo programming, the difficulty in adequately monitoring the audio would increase. For that reason, it was decided that single channel control rooms would be preferrable to the multi-channel control rooms of the past.

There would still be four main output channels and four secondary channels as in the past; however, each main channel and its secondary channel would be monitored from one of four separate automated control rooms.

Other daily functions of the Tech Center support the National Program Service. For instance, in order to ensure the consistent quality of the schedule assembled from program submissions from producing public stations and independent producers, it is necessary for each tape to be technically evaluated. Cassette dubs are made at the time of evaluation for program content screeners. Other cassette copies are made for submission to member stations who do not have access to the satellite feed, the National Captioning Institute, and to various in-house users. It is also necessary to dub a captioned version of most programs using the uncaptioned master from the producer and a caption diskette.

All of the above functions as well as the DACS message service, the insertion of data into the VBI, and the support of occasional commercial feeds were considered in the design of the system.

New requirements include the increased capability to edit promos, to repair impaired programs in-house and the ability to distribute and monitor a specialized training service to corporate customers.

Design Goals

Chief among the design goals was to have a highly integrated and automated plant where major systems worked together under computer control. It was that feature of the original plant that enabled us to stay on the air as long as we did on the night of the fire. Experience also taught that a single supplier was necessary to ensure the proper interaction among the major system components (routing switcher, machine control, master control switchers, and automation) of the plant, and that those components should be standard products as opposed to non-supportable custom items.

The routing switcher matrix size was determined to be on the order of 100 x 100 expandable to approximately 130 x 130 with video and three levels of audio all separately controllable. A variety of control panels encompassing X/Y control to single bus control with programmable features were required.

The machine control system would have to control a minimum of 30 machines of all types with expandability to at least 50 machines. Distributed control was needed to avoid single point failures. Serial communication to the machines with an "S.M.P.T.E. like" protocol using all controllable functions was also required.

The master control switching function was expected to use the crosspoints of the routing switcher with the routing switcher output buses serving as program lines. This would facilitate the automation of the entire program distribution operation.

The search for such a system led to the several recognized leaders among broadcast equipment manufacturers.

Major System Component Selection

The Robert Bosch Corporation, now BTS, was chosen to supply the major system components. Their product line closely matched the required system. The TVS/TAS-2000 Routing Switcher is modular and expandable. Its specs are among the best the industry has to offer. The TCS-1 Machine Control System provides multiple control points, serial protocol with all controllable functions supported, and expansion of control points and machines. At the time of selection, the automation system was under development but was proceeding along a timetable with design goals commensurate with our plans. The opportunity to influence the design process was an added confidence builder that the design would be appropriate to our needs. Also in our estimation, the MCS-2000 Master Control Switcher represented a significant innovation in master control switching design. The switcher's inputs are the inputs to the routing switcher with digital control of all functions and the ability to store pre-arranged panel configurations. Since the MCS-2000 was also designed to work with the machine control system, automating that switcher also automated the selection of crosspoints of the routing switcher and the functions of tape machine operation.

System Description

The Equipment Center houses all central distribution equipment. The routing switcher is at the heart of the system. The Video and Audio 1 and 2 matrices are 90 X 100 expandable to 130 X 110. The

Audio 3 matrix is 40 X 40 expandable to 130 X 40. The routing switcher has redundant power supplies in each card frame and redundant control cards. The frame containing the control cards has separate power cords plugged into separate circuits for each of the power supply modules. The software of the control cards has the "down-loadable mnemonics" option which enables us to configure the matrix from an external terminal. There is an assortment of X/Y and single bus control panels most of which permit independent control of all levels. Communication between the matrix and the control panels is via coax party line.

Other equipment within the equipment center includes: a custom 25 station Ward-Beck intercom system with group call functions, a Leitch master clock system and proc amps, a Grass Valley sync system and video DAs, Tektronix test signal generators, Volumax limiters, DATE encoders and decoders, Farinon FM subcarrier modulators and demodulators, and Collins microwave radio transmitters and receivers.

Each of the four control rooms is based around an MCS-2000 Master Control Switcher. The switcher has control of five output buses of the routing switcher. These are used as two main switching buses, two key buses and an auxiliary bus. The output of the MCS-2000 is the primary "A Channel" output feed: a secondary, "B Channel" output from each control room is a dedicated routing switcher output bus. Four control rooms were chosen instead of the two multi-channel control room configuration of the previous plant to facilitate the critical monitoring of the increased hours of stereo programming. A BTS-Bosch 10 X 1 switcher whose inputs come directly from tape machines is used for emergency back up of the MCS-2000.

Due to the diverse makeup of the many receiving stations, it is necessary for PBS to distribute both mono and stereo signals over the satellite system and to monitor both signals throughout the plant. The problem of simultaneous handling of mono and stereo feeds was solved by a creative application of the BTS equipment. When airing a stereo program, we feed the stereo signal to the MCS-2000 and create the mono mix within the switcher. The fact that the MCS-2000 can be so completely reconfigured under software control makes it all possible. The buses that feed the MCS-2000 carry the left and right signals which are handled by the main audio output cards of the switcher. A third, aux/3, output card was purchased and is wired to receive both the left and right signals that feed the main channels. The operator need only set the aux channel control to mix the two signals--an operation that can be automated or that requires only two key strokes. The outputs of the main channels are directed (again through the routing switcher) to the DATE encoders while the output of the aux/3 card is sent (also via the router) to the FM subcarrier modulator. In emergency backup mode the backup 10 X 1 switcher is also wired to provide left, right, and mono outputs from left and right inputs.

The automation system software runs on a Hewlett Packard 310 computer with 2 Mbytes of RAM, touch-screen color display, 40 Mbytes of hard disk storage and a 3.5" floppy disk drive. A separate automation computer switches the schedule in each control room. The automation software controls the main schedule and all associated machine events as well as the switching of an auxiliary schedule on the B channel bus or automation of off-air recordings through its control of the router. A routine prompts the operator to load tapes or to put a machine into remote. All manual actions taken on the master control switcher are logged through the automation computer. Also because of the processing power inherent in the computer, it is possible to run other computer functions while the software maintains the schedule.

The main videotape area has support positions for 20 Ampex VPR-3s arranged in four groups of five outside of each control room. This is to enable a single operator, in the future, to have complete responsibility for loading, airing, and monitoring a schedule. Two separate isolated rooms are set up each with a single VPR-3 and monitoring equipment for technical evaluations. These screening rooms also have 1/2" and 3/4" dubbing capability so that cassettes can be made along with the screening process. High quality monitors and small format cassette machines are distributed about the main tape area for additional screening and dubbing capability. Two Ampex AVR-1 quad machines (one equipped for double system operation) are positioned outside of the screening rooms for transfer of archival material.

Two fully equipped monitoring areas were established to support several operational functions. The Duty Supervisor's office is the central monitoring point and the final check on quality control. Incoming remote feeds are monitored and proc amps and frame syncs are controlled from the Transmission (TX) Area.

Many thanks go to A.F. Associates for their efforts in detail design and implementation of the technical facility under a very aggressive time line.

Elsewhere on the Third Floor is a sophisticated Edit Suite comprised of a single on-line room and up to three off-line rooms with a specialized graphics area. The on-line room has a Sony BVE-5000 editor, three BVH-2000 1" tape machines, a BVU-800/820 pair of 3/4" cassette machines, a Grass Valley 100 switcher, a Texta character generator, and an Abekas digital effects unit. The Edit Suite also relies on the main routing switcher for distribution. It provides us the ability to create high-quality promos for national programs and to correct the impairments of submitted tapes more quickly and cost effectively than returning tapes to the producers.

Future Consideration

Plans for the future include the addition of a Teleconference Studio using field production quality CCD cameras with a compact switching and audio package. A small format based automated time zone delay system is also under consideration. There is designed-in flexibility and expansion to accommodate these and many other enhancements to the service capability of the plant.

CONCLUSION

The success of a major design effort depends on the accurate assessment of the services to be provided today and into the future. Having the input of all concerned departments has enabled PBS to design a facility using state-of-the-art technology which is appropriate to current needs and will grow and change as the needs of the company change.

FACILITY AND EQUIPMENT PURCHASE DECISION MAKING PROCESS

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ABSTRACT

This paper discusses the purchase of a broadband circularly polarized cavity backed radiator antenna for use by a channel six television and five Class C FM broadcast facilities.

Both the financial and engineering positions are presented with emphasis on the financial area.

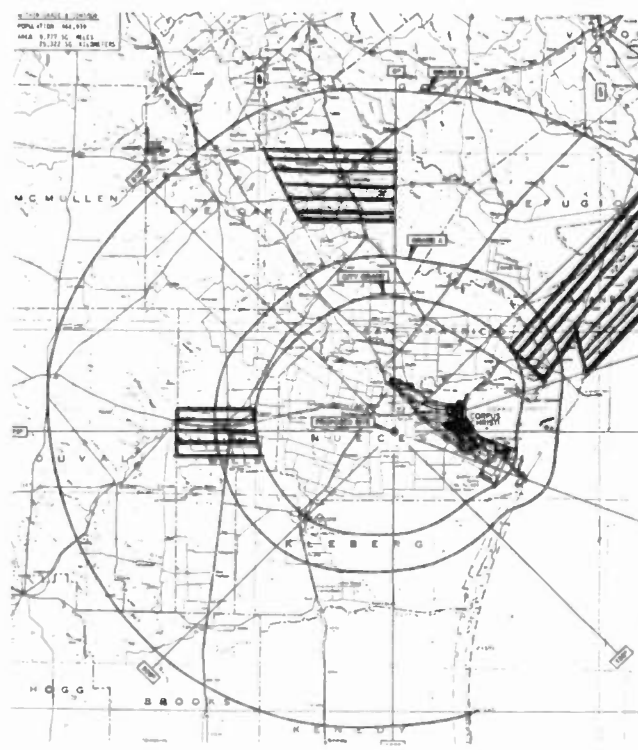
INTRODUCTION

There are three major areas in the decision making process involved with making a major equipment purchase. They are multi-faceted and lengthy. The technical requirements will eliminate many choices, while financial requirements may remove other choices. Choosing among those products remaining requires both an engineering and a financial expertise.

One example is the purchase of a new wideband television broadcast antenna. A channel six television station owner can operate a wide band antenna on his channel and share the antenna with more than half the FM broadcast band. The question this station faces is whether to purchase a narrower bandwidth antenna and only run the tv signal or purchase a wide band antenna that costs considerably more and rent this bandwidth to generate revenue.

INITIAL PLANNING

When considering the purchase of an antenna, the first two questions that arise are: (1) what is the coverage that we as a television station want? (2) What antenna/tower combination will deliver this coverage? A study of the viewing area will reveal areas of population clusters within your ADI. A study of population growth will reveal where the new homes will be built in the future. The



reference librarian at your local college can provide information on where to find this data. Plot this information on a map using different colors to show population areas and projected growth areas. Outline the whole area and add your present coverage area to complete your antenna coverage map (Exhibit 1).

Market research in our area revealed a great number of "second sets" being fed with "rabbit ears, "possibly as many as 33% of the homes with television. That same research revealed that these sets were in an urban area that we presently did not serve with an adequate signal. Our coverage map showed two new areas of predicted population growth that could be covered by a 1000 foot antenna. Being a coastal station in Texas, a great part of our signal was wasted over water and sparsely populated ranch land. Our first goal then became to cover the new population areas and reduce coverage over water and unpopulated ranch land.

A second goal was to present a strong signal to those city grade "second sets." While second sets may not show up in the "ratings books," they are important to the company from a publicity and awareness position. We wanted as many sets tuned to our station as possible and one way was have the strongest "rabbit ear" signal.

Initial Decisions

Using generic antenna and tower costs from national manufacturers as a base, a study of antenna cost vs height vs homes covered was run to find the optimum height/cost ratio. The ideal height was approximately 1000 feet.

In view of our second goal, engineering felt that a circularly polarized antenna would provide viewers with the ability to tune our signal with a minimum of "fiddling" with rabbit ears. While the cost a CP antenna is greater than a linear polarized antenna, the actual cost of a CP antenna system must include the cost of a transmitter capable of producing double the present power, increased maintenance, and additional power consumption and air conditioning costs over the life of the project. While CP was desirable, its cost for the 121 st market would be hard to justify for "rabbit ear" coverage alone.

Reducing Costs

The initial quotes from a major manufacturer were for both their standard CP pole mounted narrow band antenna and for their newer cavity backed radiator antenna. The cavity antenna had a bandwidth of 20 Mhz. One solution to the cost problem appeared obvious. Order the antenna for a low end of 82 mHz and share the additional bandwidth with the FM broadcast band. Channel six being 82 to 88 mHz. allowed FMs in the market to operate up to 102 mHz which included five present stations and two proposed.

FINANCIAL DECISION PROCESSES

Financial planners use four methods in evaluating a proposed capital purchase. They are net present value (NPV), internal rate of return (IRR), profitability index (PI), and the payback period method. The first three take into account the time value of money. By using these tools it is possible to compute a projected total cost of a project over its lifetime. In the case of the wide-band vs narrow-band antenna, the time value of money techniques suggest the more expensive antenna is the correct purchase, as the cash flows over the life of the antenna are greater than those generated by the narrow-band antenna. When making a decision based on financial analysis, the time value of money methods should point toward the same project. They will give you a sense of how much the project will really cost over the lifetime of the investment. Should the indications not favor the same project, you know you have an area of investigation to determine why the cash flow indicators do not agree.

Payback Period

The payback period method is the amount of time needed to recover the initial investment on a project. For example, if a project provides a cash flow of $ 300 per year on an investment of $1000, the payback period is 3.33 years. Using payback period, accept a project if its investment is returned within a specified period. The period is specified by management based on previous purchases, bank note payment, "gut feeling," etc. When comparing two competing projects accept the project with the shortest payback period that is less than the specified period.

While payback is simple to compute and very popular, it has a number of problems. One of the two major ones is that payback does not take into account any returns after the payback period. The project could be rejected on time only and not on the actual return on investment, if the major cash flows were near the end of the project.

Payback ignores the pattern of returns within the period. For example, two projects that have the same period of payback of the investment are considered to be equivalent. However, taking the time value of money into account, any dollar returned early is more valuable than a dollar returned near the end. Payback fails completely to account for the differences in the timing of the cash flows. Payback can be very misleading and has little to recommend it when compared to "time value of money methods".

Net Present Value

The "Net present value" (NPV) method uses the present value of each future cash flow discounted back to the present, at the appropriate discount rate (interest rate). Each present value cash flow is then "netted" (added or subtracted) to yield the total present value of the cash flows. The cost of the project is then subtracted from this cash flow total and the result is the Net Present Value. The NPV is the value that accrues to the firm from investing in that project. A positive NPV is the extraordinary return from the investment, in other words, profit. The greater the NPV, the higher the profit returned. If the NPV = 0, the project is just returning its investment cost. A NPV that is negative is not even earning its cost at the required rate of return and should be rejected. The required rate of return is a key value in NPV calculations. Also called the appropriate discount rate or interest, it is what it costs your firm to borrow money or the value lost by not investing in some other project.

Internal Rate of Return

Internal Rate of Return is the interest rate that we expect to earn on a project. It is also the interest rate that causes the net present value of a project to equal zero. When the IRR exceeds the cost of capital, a project should be acceptedandwhen the cost of capital exceeds the internal rate of return a project should be rejected. There is no quick method to compute the IRR. Calculating IRR by hand involves making an educated quess and using that quess as the cost of capital in the NPV problem. If the resulting NPV is positive, raise the estimate and recompute the problem. Keep adjusting the cost of capital until the NPV is zero. At this point, you have achieved the internal rate of return for your project. If the resulting IRR is greater than the cost of capital, you accept the project as viable. IRR alone should not be used to choose between two mutually exclusive projects.

Profitability Index

The profitability Index is a procedure to rank a firm's potential investments. The formula is (NPV/I), where NPV is the net present value and I is the investment. Since PI is a ranking procedure for a firm's total capital budget, it was used in the decision making process for the antenna purchase by ranking all of the company's capital projects for the year. The PI of each project was listed in order, highest to lowest, accepting the projects with the highest PI until the capital budget was exhausted.

FINANCIAL MODELS

While purchasing the multi-station antenna seemed obvious, some financial models were constructed to validate the assumption. For the models, the cost of capital was assumed to be 10% and the tax rate to the company was assumed to be 40%. (These figures were used in this paper because they are available in most present value and tax tables.) The first step was to do an item by item comparison of the costs of a single station conventional CP antenna and tower vs the costs of a multi-station television and FM broadcast antenna with tower. (Exhibit 2)

The land, transmission line and service fees of the two projects are similar. The cavity back antenna tower costs an additional $180,000 due to the increased wind loading, height and weight of the cavity back antenna. The building is six times larger and costs four and a half times more. The cost of a multiplexing unit is $230,00, (which is used to combine four FM stations and a channel six television station into one antenna.) The multiple station antenna itself costs an additional $234,000. The total difference between the two systems is $877,500 or $219,375 for each FM station. This is a significant savings to the FMs as the average 1000 foot FM antenna costs an estimated $1,200,000, or $10,000 per month in payments for 15 years.

Net Present Value of TV Antenna Only

(Exhibit 3) To see what the total antenna cost would be in today's dollar, a NPV is computed on the $1,089,500 cost. The present tower will be removed and sold for scrap. Subtracting demolition charges and taxes, the tower, antenna and transmission can be sold for $20,000. During the year this project was planned, a five per-cent Investment Tax Credit was in effect which would be $54,475. Subtracting these cash flows from the initial purchase price, the total cost of the project would be $1,025,025.

The additional coverage from the new antenna provides an estimated $250,000 additional television revenue each year. (This is a cash flow "IN" on the chart.) After taxes, $150,000 of the additional revenue is available to a 15 year annunity at 10 % interest. This cash flow is worth $1,140,915 in today's dollars.

The new antenna can be depreciated under The Accelerated Cost Recovery System in five years. After taxes, the present

Exhibit 2

1,000 foot, Antenna Installation
Single vs Multi Station Costs

ITEM	Single station	Multi Station
Land	$200,000 (44 acres)	$200,000
Tower (installed 1000 ft. guided)	$500,000	$680,000
Building (including electrial and Air Conditioning)	$62,500 (650 sq ft)	$288,000 (3880 sq.ft)
Antenna	$192,000 (CP pole mount)	$426,000 (CP Cavity Back)
Transmission Line (6-1/8", 50 ohm)	$115,000	$115,000
Service Fees (Attorney, Architects Consultants, surveys)	$20,000	$28,000
Diplexing Unit		$230,000
Total Project Cost	$1,089,500	$1,967,000

value of the depreciation for each year is listed. Under expenses, the old tower required $5000 per year for lights and painting or a PV of $22,818 saved over 15 years. The new tower will have no painting requirement and $3000 per year will be required for repairs and lighting. The total cash flows for the tower will be $1,487,834 in today's dollars. Subtracting Cost from Cash flows the difference is a Net Present Value of $462,809 and an Internal rate of return of 18.41 %. The single station TV antenna would generate the company $462,809 profit over the 15 year period.

Net Present Value Multi-Station Antenna

(Exhibit 4) The multi station antenna requires an additional 80 feet of tower, at a cost of $180,000. The multiplexer for four FM's and a TV costs $230,000. The building and additional service fees add another $230,000, for a total cost of $1,967,000. Each of the four FM stations will pay an initial $40,000 to buy a "port" in the multiplexer. This $160,000 contribution added to the ITC and salvage of the old tower will reduce the actual cost of the project to $1,688,650.

The monthly cash flow from the FM antenna rental will be $2000 per tenant for the first five years and then $4000 per month after year six. A decision was made to subsidize the FM stations for the first five years as a way to attract tenants for the antenna. The television revenue would remain unchanged at $250,000 per year as would the expenses related to tower maintenance. Depreciation would again be ACRS for five years. The total present value of the cash flows from the antenna for 15 years will be $2,368,334.

NET PRESENT VALUE   Single Station

Exhibit 3

Cost of capital = 10%
Tax rate = 40%

CASH FLOW	BEFORE TAX	TAX	TIME POSITION	AFTER TAX	FACTOR	PRESENT VALUE	DIRECTON
Buy New Antenna	1,089,500	1.0	1	1,089,500	1.0000	(1,089,500)	Out
Sell Old Antenna	20,000	1.0	1	20,000	1.0000	20,000	In
Investment Tax Credit	54,475	1.0	1	54,475	1.0000	54,475	In
			TOTAL COST OF PROJECT			(1,015,025)	
Sales Revnue TV	250,000	0.6	1-15	150,000	7.6061	1,140,915	In
Sales revenue Old	0	0.6	1-5	0	7.6061	0	Out
Dep Old	0	0.4	1-5	0	7.6061	0	Out
Dep New 15%	155,254	0.4	1	62,102	0.9091	56,456	In
22%	227,706	0.4	2	91,082	0.8264	75,270	In
21%	217,355	0.4	3	86,942	0.7513	65,320	In
21%	217,355	0.4	4	86,942	0.6830	59,381	In
21%	217,355	0.4	5	86,942	0.6209	53,982	In
Expense Old	5,000	0.6	1-15	3,000	7.6061	22,818	In
Expense New	3,000	0.6	1-15	1,800	7.6061	13,691	Out
				Present Value		1,487,834	

Net Present Value	472,809
Profitability Index	1.47
Internal Rate of Return	18.414

Subtracting the total cost from the total present value we find a Net Present Value of $679,684, an Internal Rate of Return equal to 17.24%, and a Profitability Index of 1.40.

Comparing the two from a financial view point, we find both would be profitable. The single TV only antenna has a positive NPV, a PI greater then one and the IRR is greater than the cost of capital.

The multi-station antenna also has a positive NPV, a PI greater than one and a IRR greater than the cost of capital. However, the IRR for the TV only system is greater than the multi-system. This discrepancy can be explained by the larger cash flows from the FM station occurring later in the project. The value of a dollar is less in 15 years than today. The NPV for the multi is larger by approximately $200,000 and this figure represents additional profit for the company.

Both antenna systems are viable investments for the company. Both have an average rate of return for the funds committed to the project. The multi-station antenna has additional good will for the company by providing a transmission medium for the FM broadcast community at a reasonable cost. Having four FM stations on one tower also eliminates additional stand alone towers in the area which reduces hazards to air navigation. Again, goodwill in the community.

CONCLUSION

There are three major areas involved in the decision making process when purchasing capital equipment (in this case an antenna.) They are engineering, financial and managerial. The managerial, decision involves the geographic areas in the ADI that need coverage. The marketing department as a department of managerial has the responsibility of selling the station to the public as well as selling

NET PRESENT VALUE MULTI-STATION ANTENNA

Exhibit 4

Cost of capital = 10%
Tax rate = 40%

CASH FLOW	BEFORE TAX	TAX	TIME PERIOD	AFTER TAX	FACTOR	PRESENT VALUE	DIRECTON
Buy New Antenna	1,967,000	1.0	1	1,967,000	1.0000	(1,967,000)	Out
Sell Old Antenna	20,000	1.0	1	20,000	1.0000	20,000	In
ITC	98,350	1.0	1	98,350	1.0000	98,350	In
Loss on Sale Old		0.4	1		1.0000	0	In
Contribution FM's	160,000	1.0	1	160,000	1.0000	160,000	In
			TOTAL COST OF PROJECT			(1,688,650)	
Rev FM's 1 to 5 yrs (2000/mo)	96,000	0.6	1-5	57,600	3.7908	218,350	IN
Rev FM's 6 to 15 yrs (4000/mo)	192,000	0.6	6-15	115,200	3.8153	439,523	IN
Additional Sales Revnue TV	250,000	0.6	1-15	150,000	7.6061	1,140,915	In
Dep New 15%	280,298	0.4	1	112,119	0.9091	101,927	In
22%	411,103	0.4	2	164,441	0.8264	135,894	IN
21%	392,417	0.4	3	156,967	0.7513	117,929	IN
21%	392,417	0.4	4	156,967	0.6830	107,208	IN
21%	392,417	0.4	5	156,967	0.6209	97,461	IN
Expense Old	5,000	0.6	1-15	3000	7.6061	22,818	In
Expense New	3,000	0.6	1-15	1800	7.6061	(13,691)	Out
				Total PV		2,368,334	

Net Present Value 679,684
Profitiability Index.................... 1.40
IRR..................................... 17.24%

commercials to the advertisers.

A "no excuses" air quality picture is a primary factor in selling your station and that picture is the responsibility of engineering. Engineering must provide its services and expertise in an efficient and thrifty manner. The antenna discussed in this paper is a prime example of an engineering product that can increase revenue to the company in a secondary manner, (by renting excessive bandwidth to the FM community.) Engineers must be alert to revenue generating devices and maximize every opportunity that presents itself. Both managerial and engineering functions directly effect the financial "bottom-line" of your company. Capital budgeting techniques and financial analysis should become a tool in every engineering manager's toolbox.

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Spreadsheets from Lotus Symphony, 1.2

NEW ACOUSTICAL MATERIALS IMPROVE BROADCAST FACILITY DESIGN

Peter D'Antonio and John H. Konnert
RPG Diffusor Systems
Largo, Maryland

The introduction of stereo TV, stereo AM radio, digital recording, compact discs, recent advances in 3D stereo imaging and emphasis on quality audio-for-video in general, has placed new importance on the acoustic design of recording and broadcast facilities. In this paper we will introduce the concept of a reflection-free zone, RFZ^{tm}, and discuss advanced acoustical designs for three environments in the broadcast audio chain- the reference monitoring or control room, the recording studio, and the announce or voiceover booth. In each of these designs we will describe the role of a new acoustical design ingredient, the reflection phase-grating sound diffusor (RPG DIFFUSOR^{tm}) and two complementary acoustical materials, a broad-bandwidth absorptive panel called the ABFFUSOR^{tm} and a new variable acoustics system based around the TRIFFUSOR^{tm} module.

INTRODUCTION

We would like to begin by making a distinction between noise control and sound control. Noise control deals with reducing or isolating competing sound or noise sources in a space. Minimizing transmission loss through partitions, by isolating massive boundary surfaces, and reducing ambient noise levels, using absorption, are two primary considerations. Sound control is concerned with the internal acoustics of a space and the effect of reflected sound on our binaural perception of music and speech. In this article we will not discuss noise control, but rather focus on interior acoustical design, in which the control of reflected sound in a room is of central importance. To achieve an appropriate fixed or variable acoustical environment there are only three basic ingredients which can be utilized- absorption, reflection and diffusion. In Fig. 1 we compare the temporal and spatial characteristics of these three design components (left column), using the energy-time curve or ETC (middle column) and the 2 KHz polar pattern (right column) determined with time-delay spectrometry (TDS). To insure that a room reproduces all frequencies equally, these acoustical surface treatments must have broad-bandwidth properties. Of the three types of surfaces, absorptive surfaces are the most well known and widely used. In fact, most people think of absorptive materials when considering acoustical treatment, because they have traditionally been used, are readily available and are relatively inexpensive. For example, the terms acoustical ceiling and absorptive ceiling have become synonymous. This is all changing. While absorptive surfaces are important, they are only one of three possible acoustical components in a balanced design. An ideal absorptive surface reflects none of the incident sound for all frequencies in the audio spectrum and for all angles of incidence. Hence, only the incident sound is measured in the ETC and no reflected polar pattern is illustrated in the top row of Fig. 1. The effectiveness of an absorptive material is characterized by the absorption coefficient, or the percentage of incident sound absorbed, as a function of frequency. Thus an absorber with an absorption coefficient of 0.95 at 4000 Hz will absorb 95% of the randomly incident sound at that frequency. It is important for the absorption to be effective over a broad range of frequencies, otherwise it will "color" the sound in the room. Thin surface treatments, 2 inches or less, of most porous absorbers, like fiberglass, sculptured foam, rockwool, etc., are very effective down to 500 Hz at which point their absorption coefficients are substantially less than the ideal value of unity. It is important to

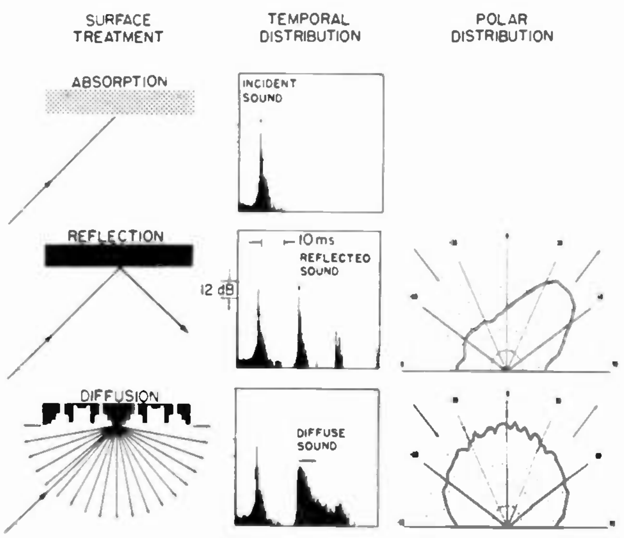


Figure 1. Comparison of the spatiotemporal properties of an absorptive, reflective and diffusive surface. The temporal distribution is depicted in the energy versus time curve or ETC (center column) and the spatial distribution is illustrated in the polar reflection pattern (right column). Arrows indicate incident and specular reflections directions.

remember that increasing the thickness and moving the material away from a boundary surface will increase low frequency absorption. Generally, for good broad-bandwidth absorption 4 inches or more are required. Random incidence absorption coefficients are useful for decay time calculations, but it is also important to determine the absorption properties for non-random incidence. That is, in acoustical design we also need to know what the absorption coefficients are for a particular angle of incidence, e.g. the angle at which sound from the monitor speaker strikes a wall. At glancing incidence some absorbers become reflective at high frequencies! This information is not available in the literature, but can be obtained by qualified acoustical consultants utilizing TDS.

An ideal reflective surface scatters all of the incident sound in a specular direction, where the angle of incidence equals the angle of reflection, for a broad-bandwidth of frequencies. This is similar to light scattering by a mirror. The reflective polar pattern is shown in the center row of Fig. 1, where the angle of incidence (-45^0) and reflection ($+45^0$) are indicated by arrows. The temporal distribution of the incident and reflected sound, which are separated by 18 ms, are illustrated in the ETC. Since the reflective surface is flat and contains no surface variation, it introduces minimal time spread. The direct and reflected sound are essentially equal in time spread and differ by only 3 dB in amplitude, in this experiment. Since the reflected energy is comparable to that of the incident sound, constructive and destructive interference occurs when they combine. This results in frequency coloration called "comb filtering". This interference can be minimized by decreasing the reflected energy using absorption or diffusion. Intense specular reflections also degrade stereo images, cause acoustic feedback and intelligibility loss. Specular surfaces, like flat or curved walls, will reflect only those frequencies whose wavelengths are smaller than the dimensions of the reflecting surface. Therefore, broad-bandwidth reflection control, which extends to low frequencies, requires large surfaces. A frequency of 50 Hz has a wavelength of 22.6 feet.

An ideal diffusive surface distributes incident sound over an appreciable time period and scatters sound arriving from any direction, uniformly into all directions, for a broad range of frequencies. The ETC in Fig. 1 (bottom row) depicts the appreciable temporal distribution of the scattered sound and the polar pattern illustrates the uniform wide angle scattering coverage of nearly 180^0, for an angle of incidence of -45^0. Due to the spatiotemporal properties of the diffusor used in these measurements the energy of each diffuse reflection is 14 dB below the corresponding specular reflection in Fig. 1. This reduces the interference when direct and indirectly reflected diffuse sound combine in a normal listening situation. When evaluating a diffusive surface one needs to consider the time distribution of the reflected energy, the spatial coverage for all angles of incidence and the range of frequencies over which the energy is scattered uniformly. The importance of diffusion has been realized since the beginning of architectural acoustics and diffusion has been approximated by cylindrical surfaces, irregular geometrical shapes, columns, statuary, egg cartons, etc. These surfaces provide useful scattering to varying degrees, but cannot simultaneously provide the broad-bandwidth spatiotemporal properties of ideal diffusion. One might wonder whether there is an optimum surface variation which provides ideal diffusion. The answer is the reflection phase-grating, RPG^{tm}, shown in Fig. 2, and the secret behind the nature of this surface lies in the paradigm of pure mathematics-number theory. The one-dimensional RPG consists of a periodic grouping of an array of wells of equal width, but different depths, separated by thin dividers. The depths are determined by mathematical number theory sequences which insure that sound arriving from any direction is scattered in all directions, for any desired frequency bandwidth. Quadratic-residue sequences, which the great mathematician Carl Friedrich Gauss played with as a child, are one example, and a diffusor based on this sequence is referred to as a QRD^{tm}. The diffraction directions for each frequency are determined by the dimension of the repeat unit and the intensity in any direction is determined by the depth sequence within a period. This type of material is completely new to acous-

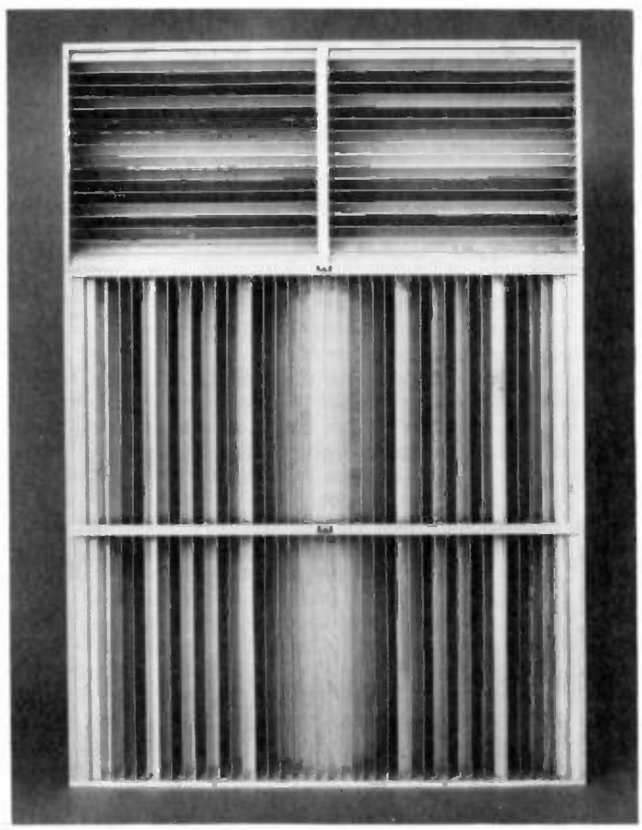


Figure 2. A cluster of three RPG diffusors. The lower two modular units provide horizontal diffusion and the upper unit provides vertical diffusion.

tical design. The RPG is the acoustical analog of the diffraction grating which has played a prominent role in optics for over 100 years. A surface of this nature was not used in architectural acoustics, until the discovery by Manfred R. Schroeder [1] of the usefulness of number theory in acoustics and the development at RPG Diffusor Systems of the RPG [2-4], over the past decade. Figuratively speaking, the RPG uniformly diffuses reflected sound the way frosted glass diffuses transmitted light or a "frosted mirror" would scatter reflected light. In addition to increasing the spatial impression in large rooms and eliminating common acoustical problems such as flutter and slap echo, resonances and frequency coloration, the RPG improves smaller "dead" sounding rooms by psychoacoustically creating the "open" impression and natural ambiance of a large room in a physically small space.

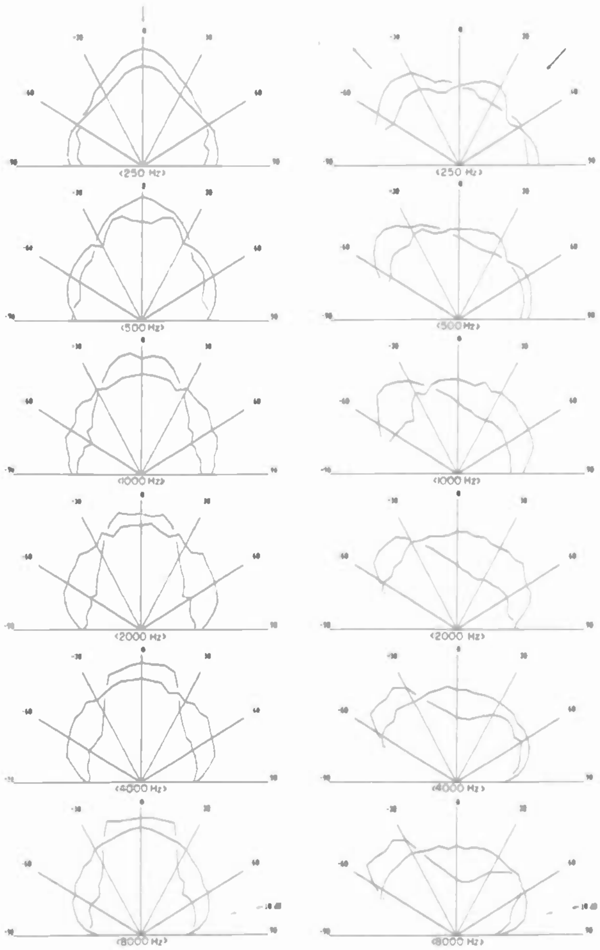


Figure 3. Overlay comparison of the experimental octave-averaged, < >, polar energy response between a QRD surface and a flat panel for 0^0 (left) and 45^0 (right) incidence. Arrows indicate the incident and specular directions. The flat panel exhibits a maximum at approximately 55 dB in the specular direction for 0^0 and 45^0 incidence. the QRD shows a uniform wide-angle broad-bandwidth response covering nearly 5 musical octaves, which is independent of the angle of incidence. The specular response is broken where it crosses the QRD responsq. At 45^0 incidence the lobing in the direction of incidence for the flat panel at 4000 Hz and 8000 Hz arises from diffraction off the leading edge of the finite flat panel and support, Data between -60^0 and -90^0 were inaccessible in data collection at 45^0 incidence.

The importance of time distribution of the reflected energy is often overlooked. A cylindrical column or irregular geometrical shape, for example, provides useful scattering but may give rise to inappropriately spaced reflections which combine to cause severe frequency domain anomalies. The RPG by its very nature provides an optimum temporal distribution yielding dense uniformly-distributed irregularly-spaced frequency notches characteristic of a truly diffuse sound field. A vertical cylindrical surface will provide uniform spatial coverage in the horizontal plane, but only at normal incidence (broadside) and for a restricted range of frequencies. The RPG with vertical wells uniformly diffuses sound into a horizontal hemidisc for any angle of incidence in the plane perpendicular to the well direction. For sound arriving from above or below this plane the diffuse hemidisc is specularly directed below or above, respectively. With respect to the range of frequencies which are uniformly scattered, the RPG is unparalleled. The QRD-4311 for example, shown in Fig. 2, provides uniform wide-angle coverage over 5 musical octaves! Experimental polar patterns for angles of incidence of 0^0 and 45^0 are shown in Fig. 3 for a QRD surface and an equivalent flat specular surface. These measurements were determined by D'Antonio and Konnert [3, #2295] in a systematic study measuring the time, frequency and directivity energy response of sound diffusing surfaces using a new boundary measurement technique based on TDS.

When used in conjunction with other types of acoustical surfaces, the RPG can be used to create

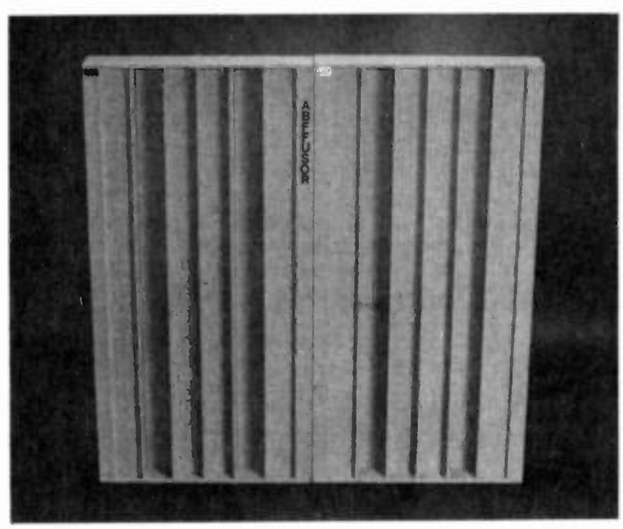


Figure 4. Photo of an ABFFUSOR, a broad-bandwidth absorber panel which can be mounted on a wall or in a standard T-bar ceiling grid.

a Complete Acoustical Treatment System. A new acoustical material called the ABFFUSOR, is shown in Fig. 4. It combines the RPG number-theory surface variation, with porous absorption and diaphragmatic panel resonance, to provide a novel broad-bandwidth absorber, with an NRC=1.0, for an average thickness of 2". It also provides a surface variation which enables broad-bandwidth absorption even when the incident sound arrives at glancing incidence. A plot of the absorption coefficients, for random incidence, of the ABFFUSOR, fiberglass, and sculptured foam are compared in Fig. 5. The ABFFUSOR effectively improves the low frequency properties of an equivalent thickness of porous material. Note the "near-ideal" absorptive properties of the ABFFUSOR. The absorption coefficient even at 125 Hz is 0.82!

Flexibility in the acoustical environment is becoming more important to suit space restrictions and the various requirements of a working facility. Dialogue one day, small musical group another, vocalist another, etc. With variable acoustics in mind the TRIFFUSOR was developed. The TRIFFUSOR is a rotatable triangular column containing a diffusive, absorptive and reflective side. TRIFFUSOR modules can be mounted side-by-side in a linear array, Fig. 6. A wide range of acoustical environments can be created by appropriate surface selection.

The RPG Diffusor System represents essential acoustical building blocks which were formerly unavailable. Consequently, it is having a profound influence on architectural acoustics and is finding widespread application in all critical listening and performing environments including: the audio/video recording/broadcast industry, performing arts facilities, churches and religious broadcasting, schools, rehearsal spaces, auditoriums, audio/video and teleconferencing facilities, entertainment venues, and home audiophile use. A few broadcast installations include:

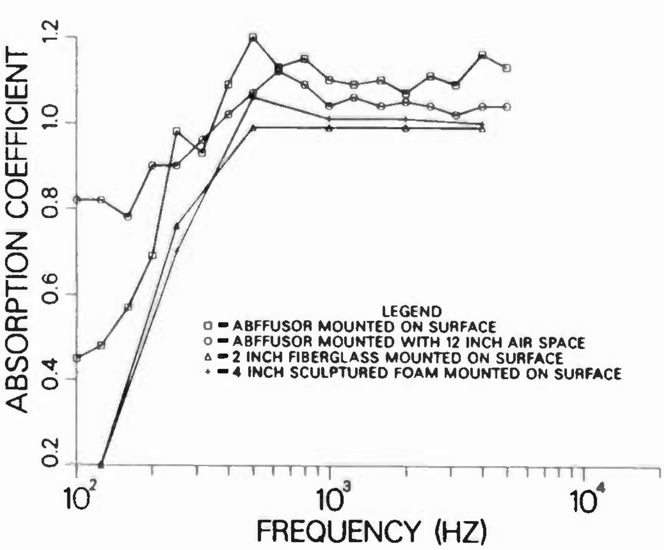


Figure 5. Comparison of the absorptive properties of the ABFFUSOR, 2" of fiberglass and 4" of sculptured foam.

Academy of Music, WFMT, Philadelphia Orchestra; BBC Malda Vale 4, London, UK; Jimmy Swaggart Ministries, Baton Rouge, LA; Johns-Hopkins University Public Radio, WJHU, Baltimore, MD; KMJQ, Houston, TX; National Public Radio, Washington, DC; NBC-TV (Bill Cosby Show, Saturday Night Live and David Letterman); NBC Burbank Post Production Studios; NBC Stereo TV Affiliate Training Center, New York; Prairie Public Radio, Fargo, ND; Radio New Zealand, Wellington; Rogers Radio, Toronto, Canada; University of Missouri, KWMU; WBGO, Newark, NJ; WFMT production room and live broadcast studio, Chicago, IL; W-Lite 95, Studio 95, Cincinnati, OH; WNKS, Columbus, GA; Word of Faith World Outreach Center, Dallas, TX; WTIC-TV, Hartford, CT.

BROADCAST FACILITY DESIGN

In general room dimensional ratios are selected first for optimum modal frequency distribution. Boundary surfaces are then angled for appropriate reflection control and diffusive surfaces are positioned to create a diffuse sound field. Absorptive surfaces should be added last, to control primary boundary reflections and adjust the decay time. Since absorption has traditionally been the primary acoustical treatment it sometimes has been overused. Some acoustical problems for which absorption has been customarily specified can also be treated with diffusion. One example is the control of a flutter or intense specular reflection (slap echo). Absorption will minimize the problem, but energy is removed with the potential acoustical side-effect of creating a "dead" space. This is all too common in broadcast studios, where the credo sometimes followed was "Deader is Better" or "No Acoustics Is Better Than

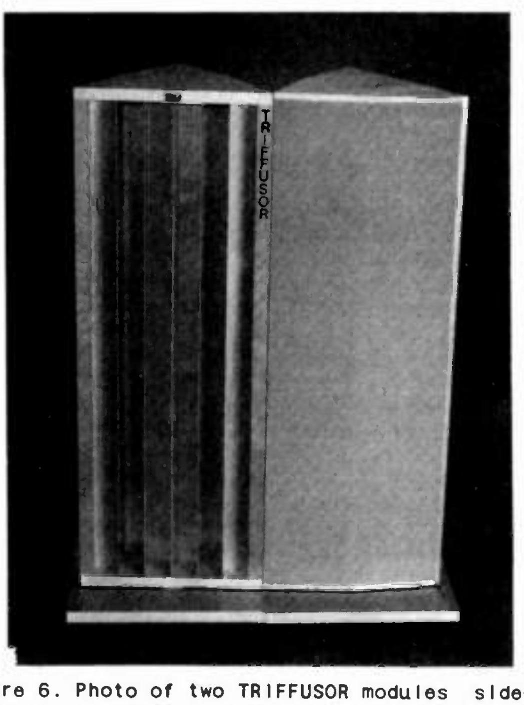


Figure 6. Photo of two TRIFFUSOR modules side-by-side with a diffusive and absorptive side facing forward. The reflective side is not visible.

Bad Acoustics". Diffusion can also be used to control an intense specular reflection, without absorption, by uniformly scattering the sound so that the energy in any one direction is minimized. In Fig. 7 we see how a 16 sq. ft. diffusive panel 8" thick redistributes the energy of an intense echo by diffusion. Thus we can remove the problem and maintain a natural ambiance. These new acoustical materials or building blocks can help designers to provide effective versatile designs easily, economically, efficiently and esthetically.

Recording Control Room

Our research has focused on effective ways to optimize both LEDE^{tm} [5] and conventional designs, by implementing an RFZ over a wide area surrounding the mix position and creating a dense diffuse sound field having significant lateral components in the room, using RPG diffusors. The RFZ is achieved by splaying, massive speaker boundary surfaces, which can contain distributed broadbandwidth absorption similar to the ABFFUSOR, thereby minimizing boundary reflections at the mix position. The RFZ permits the accurate binaural perception of pre-encoded spatial textures over a wide area, minimizes speaker-boundary interference frequency coloration, caused by very early reflections, and allows the formation of an initial time delay gap (ITD) before the onset of indirect reflected energy. The diffuse sound field is created using RPG diffusors on the rear walls. The creation of an ITD with the RFZ allows the indirect energy reflection pattern to be sequenced at any arrival time desired, and directionalized with significant lateral components derived by RPG orientation. Efficient coupling between specular surfaces on the walls, floor and ceiling and diffusive surfaces is critical in providing a uniformly dense reflection pattern throughout the decay time of the room. Low frequency modal response is optimized through the use of a new low frequency diffusion system (LFD^{tm}) and/or low-frequency absorption.

In Fig. 8(top) and 8(bottom) we show a plan view and elevation of a typical control room utilizing

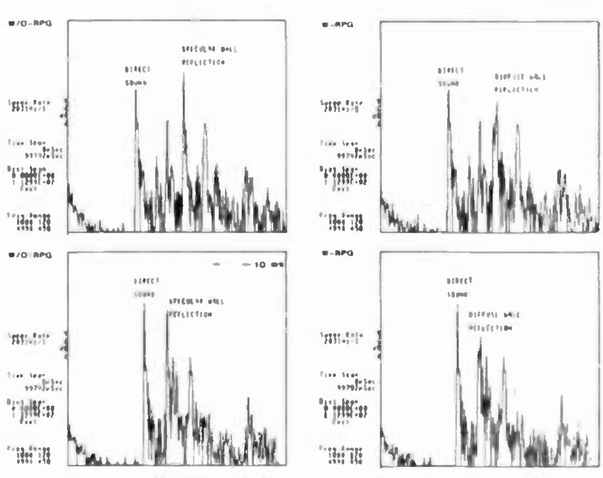


Figure 7. The ETC illustrates how diffusion can be used to redistribute the energy of an intense specular reflection. The upper measurements were made without (W/O-RPG) and with RPGs (W-RPG) present at a listening position 15' from the wall. The lower measurements were made 6' from the wall. Note that the energy of the diffuse wall reflections is decreased approximately 6 dB by diffusing the energy uniformly into the sound field, instead of absorbing it.

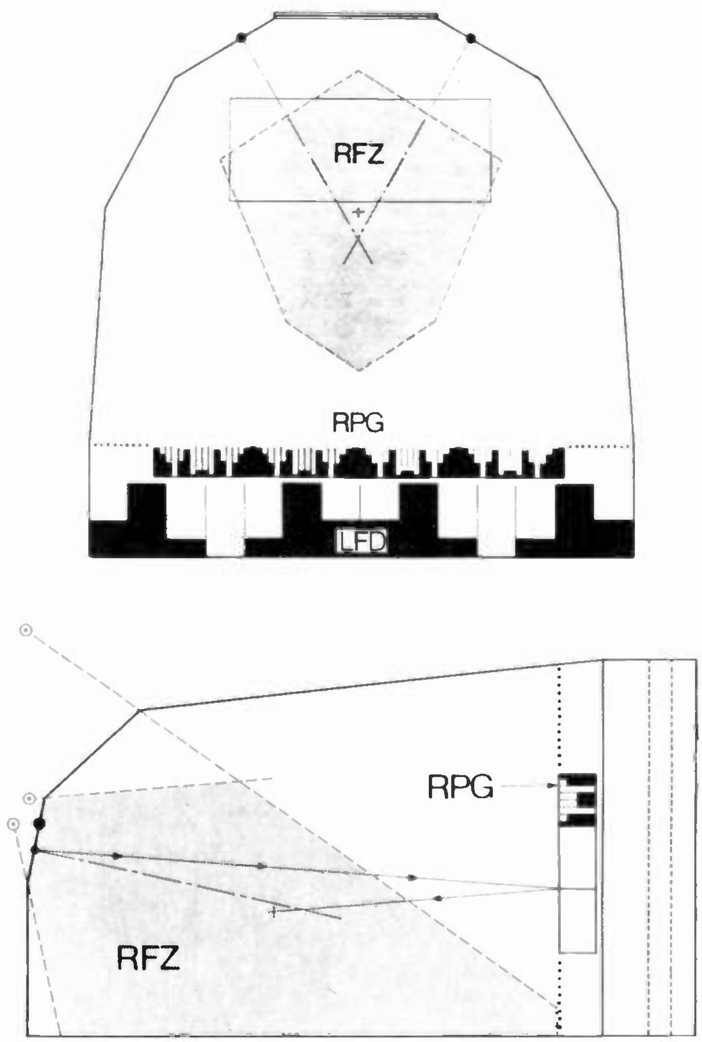


Figure 8. (Top) Plan view of an RFZ/RPG control room with LFD. Limiting reflections (dashed lines) of the speakers (large dots) from surface boundaries form a symmetrical six-sided RFZ (shaded), extending across the width of the console. (Bottom) Elevation showing virtual sources (open circles) for woofer (large dot) from front wall, forward sloping ceiling baffle and ceiling, which is shown along with tweeter orientation (long-short dash). Dotted lines indicate fabric or light absorptive treatment.

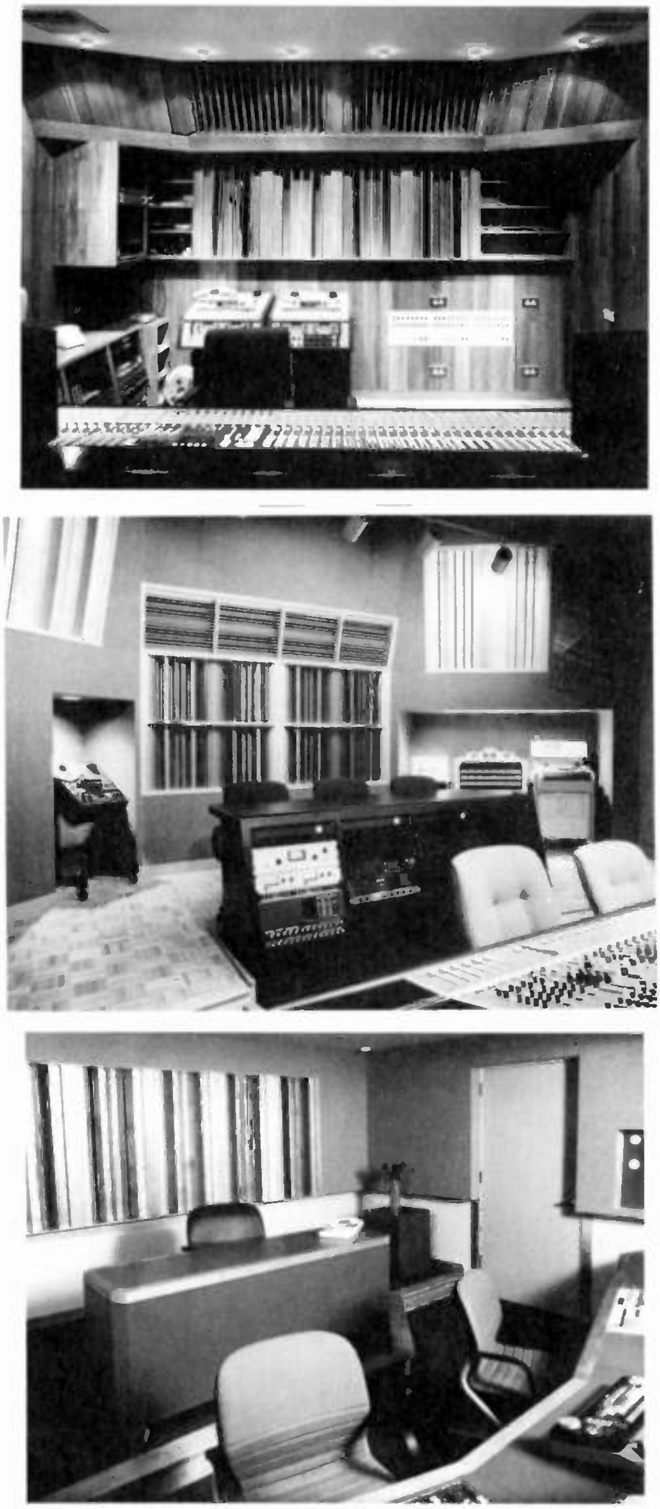


Figure 9. View of the RPG diffused rear wall in RFZ/RPG control rooms at (Top) Red Bus, London, UK. Acoustical consultant Neil Grant, DRL London. (Middle) Welk Music Group, Nashville, TN. Acoustical consultant Bob Todrank, Valley Audio. and (Bottom) San Francisco Production Group, San Francisco, CA. Acoustical consultant Randy Sparks, RLS Acoustics.

the RFZ, RPG and LFD. Fig. 8 illustrates one of many designs to create a reflection-free zone (shaded regions) surrounding the listening position (+), and a broad-bandwidth diffuse sound field. Three RFZ/RPG installations are shown in Fig. 9. An ETC of a typical small broadcast control room before and after treatment is shown in Fig. 10. Reflections in the untreated room (top) are numbered. Reflection 1 is due to the floor and 2 the ceiling between speaker and listening position. Reflections 3-10 occur from side and rear surfaces. After an RFZ/RPG acoustical treatment (bottom), the room takes on near-ideal characteristics. The RFZ establishes a clean ITD 24 dB below the direct sound and the RPGs establish a

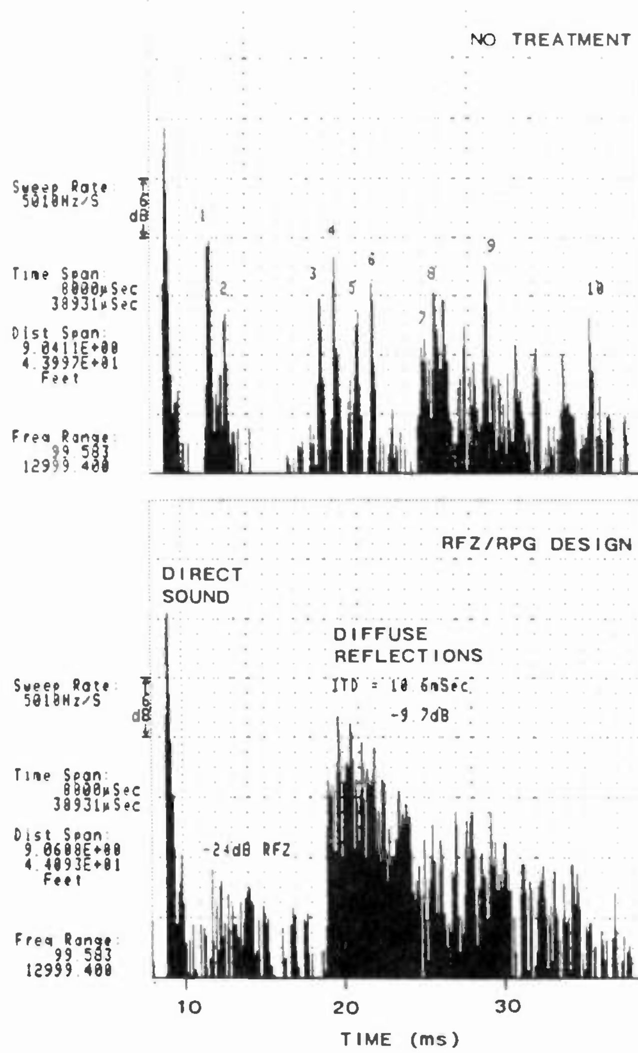


Figure 10. Top- ETC of a typical small broadcast studio before acoustical treatment. Note the interfering reflections in ITD (1,2), the sparse reflection pattern (3-10) and intense late reflections (8-10). Numbered reflections are explained in text. Bottom- ETC after a typical RFZ/RPG acoustical treatment. Note the clean ITD between the direct sound and the exponentially-decaying dense diffuse reflections. Courtesy of Charles Biliello Associates, W. Hempstead, NY.

dense and diffuse exponentially-decaying sound field, without any late interfering reflections (8-10) present before treatment.

These control rooms are also useful for "open-mic" production rooms or "talk-show" formats since the diffuse sound field minimizes the electro-acoustic interference of intense specular reflections.

Figure 11. Suspended ceiling mounted RPG diffusors at (Top) Blue Jay Recording Studios, Carlisle, MA. Acoustical consultant Russell E. Berger. (Bottom) SARM Recording Studios, London, UK. Acoustical consultant Neil Grant, DRL London. Ceiling systems integrate diffusion, absorption, and lighting.

Recording Studio

The RPG can be used in three ways in the recording studio:

1. Provide a uniformly diffuse sound field throughout the room, with an exponentially decaying sound field devoid of flutter, late slap echos and other discontinuities. To accomplish this the RPG can be mounted on the walls and/or in standard ceiling T-bar suspended systems. This arrangement provides a spacious sound with the sweetening of a concert hall. When ceiling mounted, the RPG can be used to provide an integrated ceiling system of diffusion, absorption, lighting and HVAC. In addition, much of the necessary electrical, plumbing, and HVAC supply lines can be easily hidden by this cost-effective technique, leaving them accessible for maintenance. An integrated ceiling system can be seen at Blue Jay Recording Studios, Carlisle, MA in Fig. 11(top) and SARM Recording Studios, London, UK in Fig. 11(bottom).
2. The RPG can be used to provide fixed or variable localized diffusion. Fixed diffusive elements can be installed in a section of a large multiple use room which is set aside for particular usage. Ceiling clouds or localized wall systems can be used. Variable localized acoustics can be achieved using the rotatable TRIFFUSOR modules which contain a diffusive, reflective and absorptive side. A linear array can be mounted into a wall element or in a movable variable acoustics panel. A recording arrangement which is particularly effective is to use a figure "8" pattern and position the microphone between the sound source and the variable RPG TRIFFUSOR cluster. The mic will pick up both the direct sound followed in time by whatever sound is returned from the TRIFFUSOR.
3. The RPG ENSEMBLE^{tm} can be used as an acoustical shell for orchestral or choral group recording to provide broad-bandwidth diffuse reflections and heighten the sense of ensemble for performing musicians, enabling them to hear themselves and other performers. The RPG is an ideal ensemble reflector because it provides an optimum temporal pattern of reflections and uniform scattering over a wide-angle, for any desired frequency range. The arrival time and directionality of diffuse reflections can be adjusted by appropriate placement for different musical motifs.

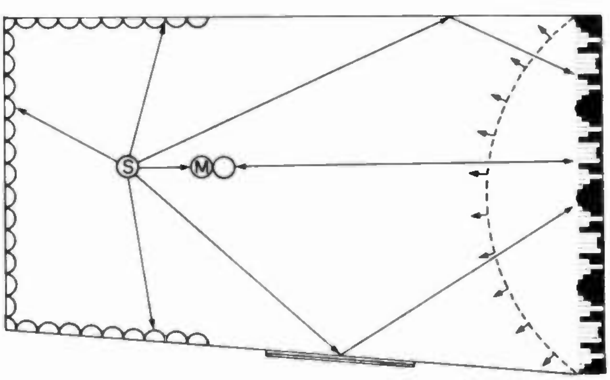
Customarily absorption has been used to reduce the acoustic disadvantages of mobile studios. The RPG offers a complementary surface treatment which provides a natural ambiance or "open" impression in this restricted space.

Voiceover Booth

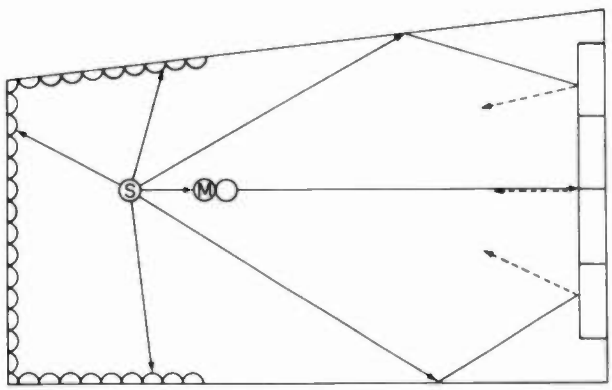
The announce booth is an important part of commercial broadcast. Because of space considerations, voiceovers are usually done in very restricted quarters. These impoverished acoustical environments translate, unfortunately, into dull recordings which lack ambiance. The problem with small rooms is that the ITD is very short, there is very little diffuse energy and modal effects cause position dependent intensity variations (hot spots and voids). By utilizing the principles of an RFZ and RPG diffusors we can now improve the acoustics of small rooms by psychoacoustically creating the impression of a larger room in a physically small space. The room should be oriented so that the talent is speaking into the longest dimension and permanent reflective items, such as windows and doors, should be situated along the side walls. They can then be advantageously utilized as delayed sound sources to the diffusors located on the farthest wall opposite the talent. Strategically placed absorption is used to create an RFZ around the microphone to establish an ITD. In Fig. 12 we show a plan and elevation of a voiceover booth which has been optimized. The RFZ is created using broad-bandwidth absorption behind and to the sides of the microphone, to minimize early reflections which provide the auditory system with clues as to the size of the room. Broad-bandwidth absorption, provided by a panel like the ABFFUSOR, is stressed so as not to accentuate the low frequency modal problems in small rooms. The farthest wall is treated with RPGs, which spatially and temporally diffuse the sound returning back to the microphone. The ceiling and side walls should be splayed slightly, 1" per foot to minimize flutter echos, and reflect the sound to and from the RPG surface, thus increasing the density of diffuse energy. Many geometries can be made to work, if the criteria of an RFZ and a sequenced diffuse sound are established. If the room does not conform to this rectangular shape and is smaller and square, then a more spacious impression may be created by using a suspended diffusive ceiling mounted in standard T-bar grids shown in Fig. 11. Modal modification can be accomplished using low frequency TRIFFUSOR modules called KORNER-KILLERS^{tm}, which contain one diffusive side and two absorptive sides and are positioned in the corners, damped membrane or slot resonator low frequency absorbers.

^{tm} DIFFUSOR, ABFFUSOR, TRIFFUSOR, RPG, QRD, RFZ, LFD, ENSEMBLE, and KORNER-KILLER are trademarks of RPG Diffusor Systems, Inc. LEDE is a trademark of Synergetic Audio Concepts.

VOICEOVER ROOM



PLAN



ELEVATION

Figure 12. Plan and elevation for a voiceover booth design which increases the apparent size of the room and enhances announcements. Absorptive treatment is indicated by semicircles, solid lines indicate direct sound and dashed lines diffuse sound, S is the source and M is a microphone with a figure "8" sensitivity pattern.

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[3] P. D'Antonio and J.H. Konnert, Audio Eng. Soc. Preprints, 2156, 2157, 2158, 2255, 2295, 2364, 2365, 2368.

[4] P. D'Antonio and J.H. Konnert, "Advanced Acoustic Design of Broadcast and Recording Facilities", NAB Engineering Conference Proceedings, TV Multichannel Sound Session, pp. 215-224 (April 1986).

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MEASURING OCCUPATIONAL RF RADIATION EXPOSURE ON BROADCAST TOWERS

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I INTRODUCTION

The problem with making power density measurements on a multiple source tower is

1. It is difficult
2. You are not making measurements from any one individual radiating antenna
3. The equipment normally used will measure the signal from the radiating antenna and all other reflecting sources
4. The results cannot be related directly to predicted signals that are presented in OST Bulletin #65

In spite of the above, we have regulations, and these regulations require that an evaluation be made. The regulations also require that the results be presented in a manner to show that you are not exceeding certain absolute limits.

In this paper we will present prediction and measurement procedures that can be used to make power density measurements. We will also evaluate the results of measurements made on two large multiple source towers.

II TYPE OF SERVICE

Of special interest are multiple source towers where many antennas are operating in different services and different frequencies. A tower may have the following radiating antenna:

VHF-TV	60 or 200 MHz
UHF-TV	400 to 800 MHz
FM	100 MHz
Mobile	80, 150, 400, 900 MHz
Microwave	7,000 - 13,000 MHz
ENG	2,000 MHz

III DEFINITION OF TERMS

True power density is the cross product ($\overline{E}X\overline{H}$) of the electric ($\overline{E}$) and magnetic ($\overline{H}$) fields at any point in space.

If we assume that we are in the far field and we have a plane wave condition, the equivalent power density can be determined by measuring the electric or magnetic field and converting it to power by the intrinsic impedance of space (120π).

$$\text{Power Density} = E^2/120\pi \quad W/m^2$$

$$\text{or} = I^2/120\pi \quad W/m^2$$

All of the regulations define the fields to be plane wave, far field, whole body, and like polarization. This is important in interpreting any measurement since many of the measurements on the tower are not in the far field nor are they plane wave. This is especially true of reradiation due to currents flowing on non-resonant metal surfaces, such as tower legs and transmission lines.

Interpreting the data is difficult since there are no regulations specifying measurement procedures. An important parameter is how far your probe should be from a radiating object. Reradiation from the tower members will decrease rapidly for FM frequencies but in some cases will increase for UHF frequencies. Tower members tend to resonate more at one frequency (FM) than others.

FAR FIELD

Space around any radiating structure can be divided into three regions.

1. Reactive Near Field - Non Radiating

   $R < \lambda/2 \qquad \lambda = \text{wavelength}$

2. Radiating Near Field

   $R < 2L^2/\lambda \qquad L = \text{longest dimension}$

3. Radiating Far Field-Plane Wave Region

   $R > 2L^2/\lambda \qquad R = \text{Distance from Source}$

All biological experiments are made with plane waves, e.g., in the radiating far field (region 3 above).

An example of non-radiating fields would be the static fields that exist inside a transmission line. If one probed the field between the inner and outer conductor of a coax line, one would measure high non-radiating fields.

Two equally spaced transmission lines running up the tower can act like a two-wire transmission line and conduct energy.

TIME EXPOSURE

The EPA* has recently specified that since exposure limits can be time averaged and are based on a six-minute period, short term exposure can be higher if the exposure time is reduced proportionally. For example:

Time	Permitted Exposure Level
6 min.	1 mW/cm^2
3 min.	2 mW/cm^2
1 min.	6 mW/cm^2
30 sec.	12 mW/cm^2
10 sec.	36 mW/cm^2

The only one on a tower should be a rigger or tower maintenance engineer. This interpretation permits procedures to be established that will allow a rigger to climb through a high RF region if it is done in less than specified time periods.

INSTRUMENTATION

The measurements made use the (a) Holaday broadband field strength meter H1 3001 or (b) Narda broadband isotropic radiating monitor, Model 8606. Both are field strength meters that measure relative field (not power) and assume that the measurement is made in the far field where a plane wave condition exists, and equivalent power density can be calculated. Many of the measurements made on the tower are in the near field and not far field.

Both of these meters are calibrated with plane waves, which, of course, implies far field. The results obtained can only be used as relative and not absolute readings of power density.

*Real Time Averaging for Determining Human RF Exposure" by R. Tell, EPA - presented at NAB 1976.

WHOLE BODY

Another term used in the regulations is "Whole Body". This is interpreted to mean that the limits can be average over the length of a man.

If the power density is plotted continually over the vertical length of the tower (in the vicinity of the ladder), readings can be averaged over 6 ft. increments.

Like polarization implies the polarization should be oriented with the length of a body. The instrumentation is not polarization discriminating.

IV TYPES OF ANTENNAS

The types of antennas to be measured would fall into three categories (see Fig. 1):

1. Side-mounted antennas
2. Panel antennas
3. Top- or pole-mounted antennas

SIDE-MOUNTED ANTENNAS

Side-mounted antennas include both FM and mobile with the FM antenna contributing most of the energy in the tower because of the high input power. The radiating center (FM) is typically 6 ft. from the tower leg placing the climbing ladder 8 ft. from the radiation center. Mobile antennas are of little concern because of their low input power and intermittent operation.

PANEL ANTENNAS

The panel antenna would include Master FM, TV (Butterfly, Zig Zag, and most of the C/P antennas), and microwave antennas.

The radiation into the tower will usually be very low (-20 dB) provided the ground plane of the panel is wrapped continuously around the tower and is one continuous ground plane in the vertical aperture. Grounding is also very important.

Existing installations were only concerned with these effects on the radiation pattern. Leg mounted UHF panels can have considerable radiation back into the tower which has little effect on the radiation pattern because of the large electrical spacing of the tower members

Microwave antennas will not present a problem because of the low input power, and the back radiation is always less than the forward gain.

TOP- OR POLE-MOUNTED ANTENNAS

This would include Bat Wings, Super Turnstyle, Slot Radiators, and Waveguide Antennas.

All of these antennas have the climbing element in the field of the radiation element and should never be climbed while operating.

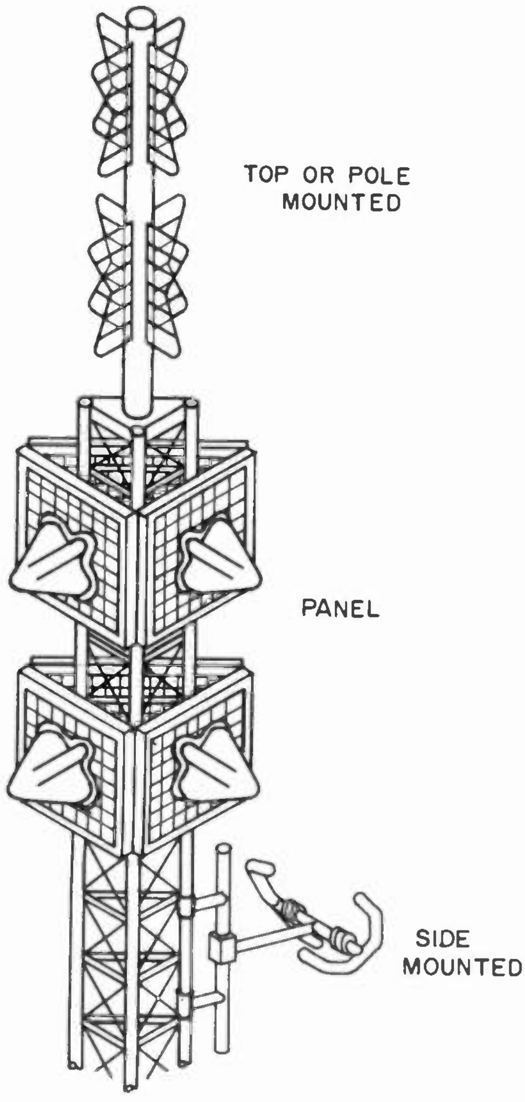


FIG. 1

V PREDICTING POWER DENSITY LEVELS

Most measurements will be made in the near field of the array but in the far field of the individual radiating element.

If we were to probe the field (see Fig. 2) at different distances from the array, we would see that the array pattern does not take shape until we are a great distance from the tower (see Table I). Therefore, power density levels within the tower can be predicted by evaluating the radiation from each radiator.

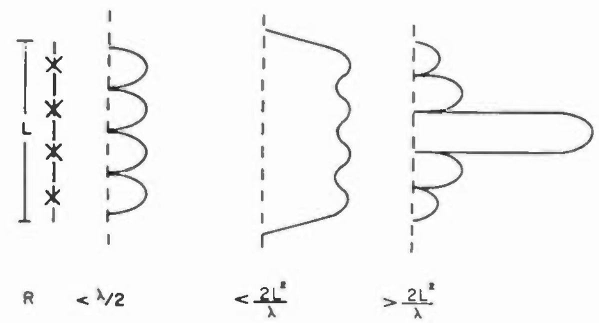


FIG. 2

DISTANCE IN FEET ALONG MAIN BEAM TO $1 mW/cm^2$ POWER DENSITY LEVEL AT MAX ERP	
Lo V/FM 100 KW	251 FT.
Hi V 316 KW	443 FT.
VHF 5.000 KW	1.764 FT.

TABLE I

If we assume that each radiator is isotropic, and we know the input power to each radiator, we can calculate the power density at any distance by calculating the radius of a sphere on whose surface there is an "X"mW for every square centimeter of surface area (see Fig. 3).

Most element radiation patterns are not isotropic but rather have a pattern like a dipole ($E=\cos^N\theta$), so the levels will be somewhat higher but the scatter in the measurement process due to the secondary radiators will be much greater than this error.

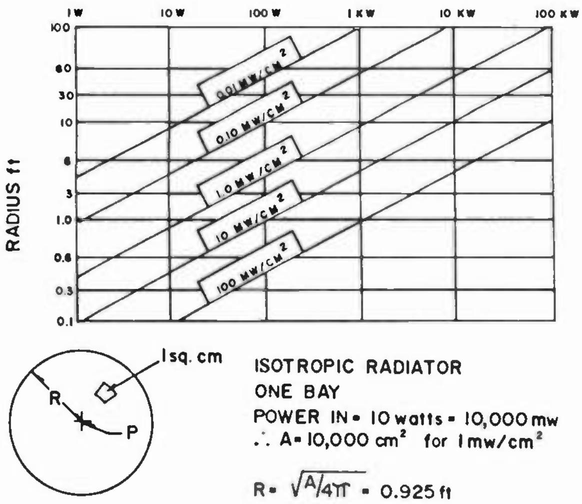


FIG. 3

An example of the calculation for a single radiator is shown in Fig. 4. If we assume an element with 20 kW input power, the radius of a sphere with "X"mW on the surface will be

Power Density	Radius
30 mW/cm²	8 ft.
10 mW/cm²	14 ft.
1 mW/cm²	40 ft.

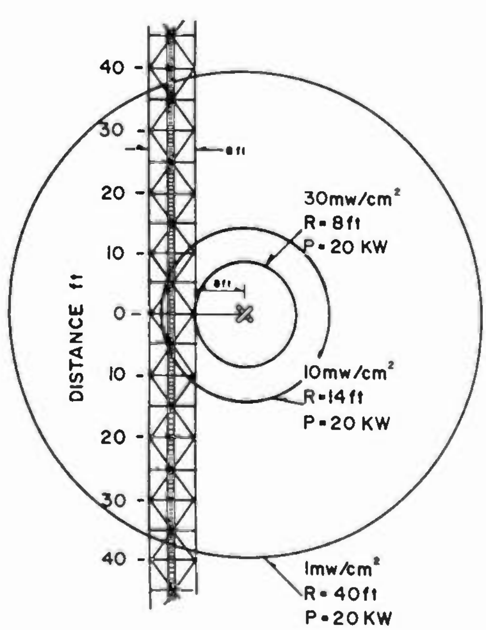


FIG. 4

VI MEASURED DATA

The enclosed data represents measurements made on two towers. Measurements were made every 5 ft. in front of the climbing ladder (see Fig. 5). The test probe was moved over an area of 1 sq. ft. and the average reading recorded.

TOWER A	TOWER B
Bat Wing (4)	Candelabra (3) UHF
Zig Zag (44)	FM-1
Super Gain (5)	FM-2
Butterfly (2)	FM-3
FM-1	FM-4
FM-2	

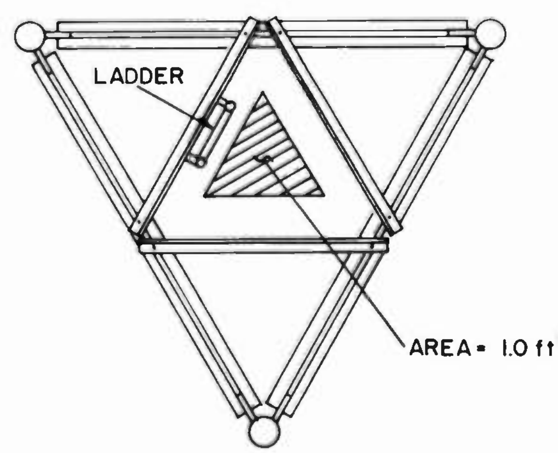


FIG. 5

Using Fig. 3 the predicted power densities were first plotted for each antenna (see Figs. 6 and 7). The results of measurements on Tower A are plotted in Fig. 8 and Tower B in Fig. 9.

The solid line for Tower A is with all antennas operating, and the dash line is with the Zig Zag and FM-2 not operating.

For Tower B, FM-4 was an auxiliary and not radiating at the time. Measurements were made up to the base of the Candelabra but not in the aperture.

Using Fig. 3 the predicted power densities were first plotted for each antenna (see Figs. 6 and 7). The results of measurements on Tower A are plotted in Fig. 8 and Tower B in Fig. 9.

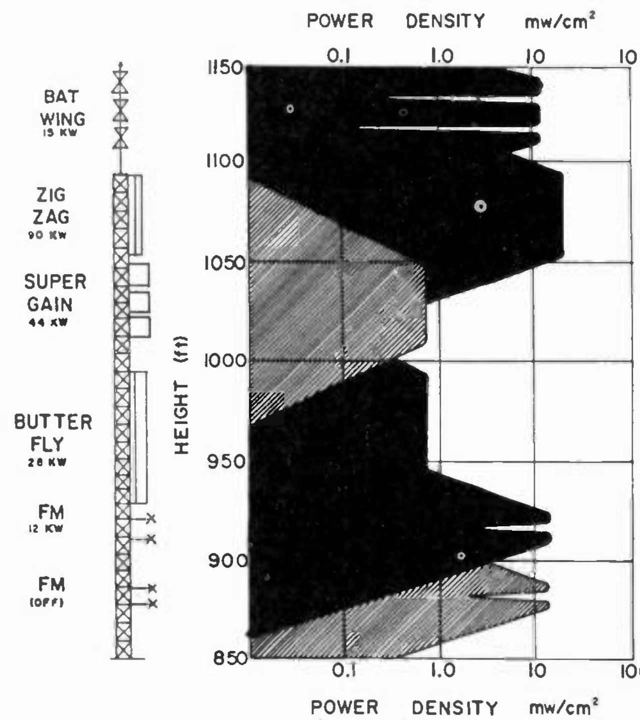
The solid line for Tower A is with all antennas operating, and the dash line is with the Zig Zag and FM-2 not operating.

For Tower B, FM-4 was an auxiliary and not radiating at the time. Measurements were made up to the base of the Candelabra but not in the aperture.

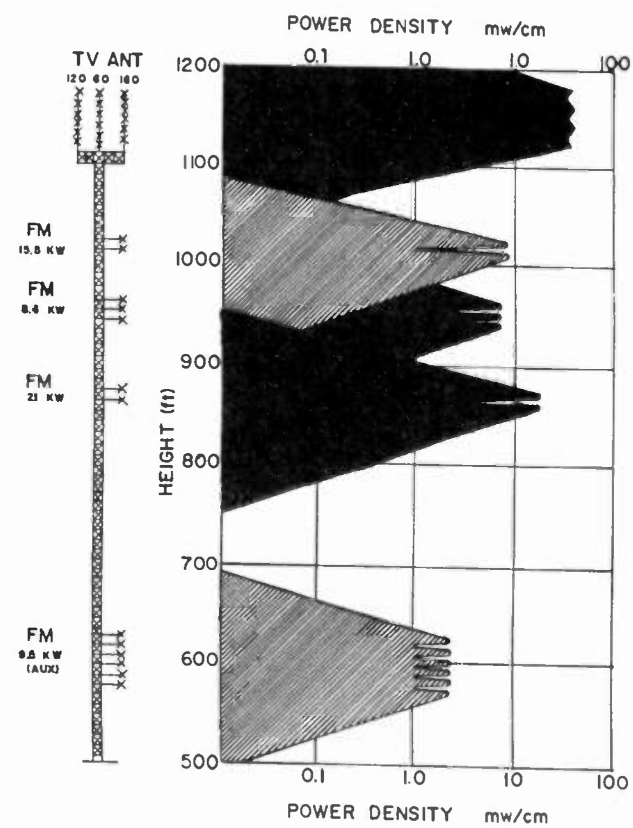
It should be noted that the measuring instrument is broadband and with such a mix of frequencies and polarizations (Tower A, 50 to 600 MHz, and Tower B, 100 to 800 MHz) at any given point on the tower. We have no way of knowing which frequency is contributing most to the recorded readings.

This is apparent in Tower A, with FM-2 OFF, the data recorded (dashed line) in back of FM-2 are probably from FM-1. Likewise, the readings in back of the Zig Zag, when OFF, are probably from the Bat Wing and the Super Gain Antenna.

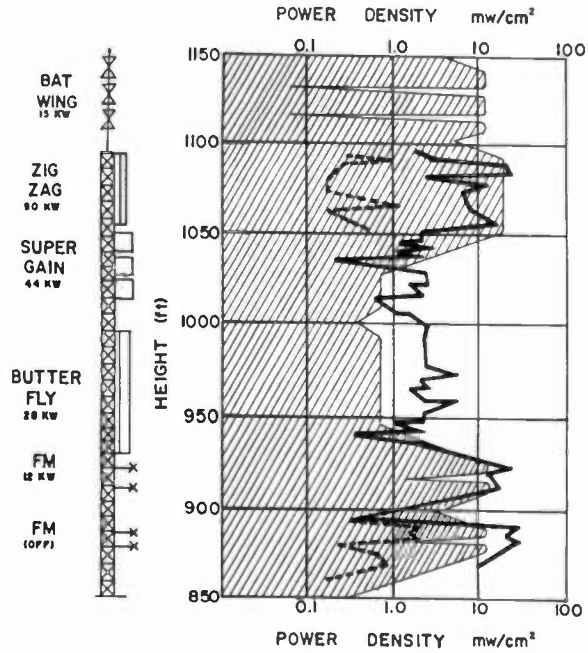
On Tower A a dual 3-1/8 line ran the full length of the tower. FM-1 energy was coupled to the two-wire transmission line (approximately 100Ω) formed by the two pipes. Although these levels are high, they are not radiating. Measurements (see Fig. 10) show that the distance between the nulls (λ/2) correspond exactly to FM1.



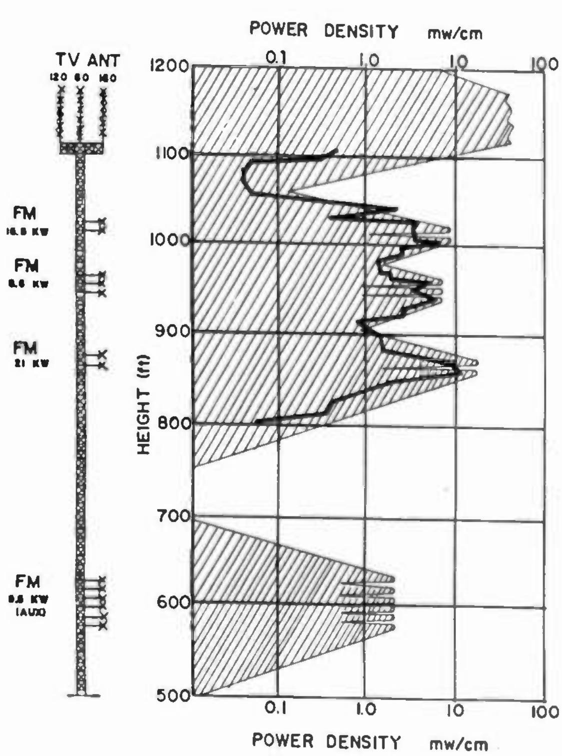
TOWER A
FIG. 6



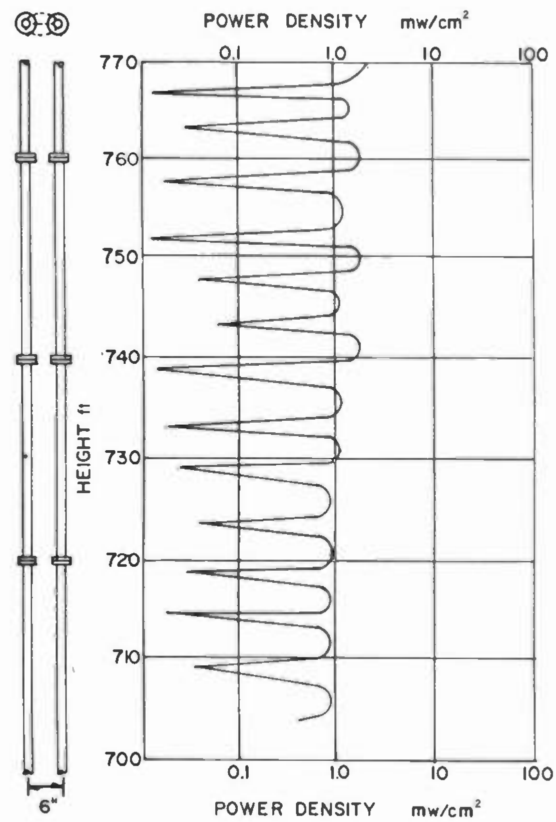
TOWER B
FIG 7



TOWER A
FIG. 8



TOWER B
FIG 9



LEVEL BETWEEN DUAL 3⅛" LINE

FIG. 10

IDENTIFYING AND MANAGING PCB'S IN BROADCAST FACILITIES

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INTRODUCTION

Much publicity has been given to the problems of polychlorinated biphenyls (PCBs) in recent years, and oddly enough, a significant amount of that publicity occurred through media which utilized products and electrical equipment containing PCBs. Such products included the ink on newspaper printed pages and the transformers or capacitors creating the broadcast transmission signals. This article will address PCB situations involving broadcast facilities, and suggest some of the things owners or managers of stations need to be aware of in order to comply with the regulations, control their risks, and minimize their liabilities.

HISTORY[1]

Polychlorinated biphenyls were introduced more than 50 years ago and manufactured by one major chemical company under the trade name Aroclor. They were used in a wide variety of applications including inks, adhesives, carbonless copy paper, hydraulic fluids, heat exchangers, and many more. Because of the superior dielectric insulating and heat transmission properties, it was an excellent fluid for use in electrical equipment. Due to its non-flammable characteristic, it became the standard accepted by the fire insurance underwriters for use in transformers installed in buildings.

The generic name for PCBs is askarel. When electrical equipment manufacturers purchased the askarel from the supplier, it was generally already mixed with a tri-chlorobenzene (TCB) solvent in some desired ratio to make it flow better or give it slightly different characteristics. This resulted in many different trade names for the askarel fluid.

Following an occurrence of illness from food/feed contamination in Japan in 1968, PCBs were thought by some to present a serious threat to human health. Early concerns about cancer risk have not been supported by scientific studies, and the principle known effect on humans is reported to be skin eruptions known as chloracne. PCB does accumulate in the food chain, is persistent and biodegrades slowly.[2] The clinical toxic rating is the same as kerosene and turpentine, which are less toxic than aspirin or tobacco when ingested in equivalent amounts.[3]

In 1983 and 1984, there were several fires involving PCBs, and investigators found that when subjected to temperatures around 600°C, both the PCB and TCB produced a by-product called polychlorinated dibenzofurans (PCDF's). The partially burned TCB also produced polychlorinated dibenzodioxins (PCDD's). It is pertinent to note that PCDF was also later found in the fluid which had made the Japanese people ill in 1968,[4] but this was not determined until long after the U.S. Congress had passed the Toxic Substance And Control Act (TSCA) in 1976.

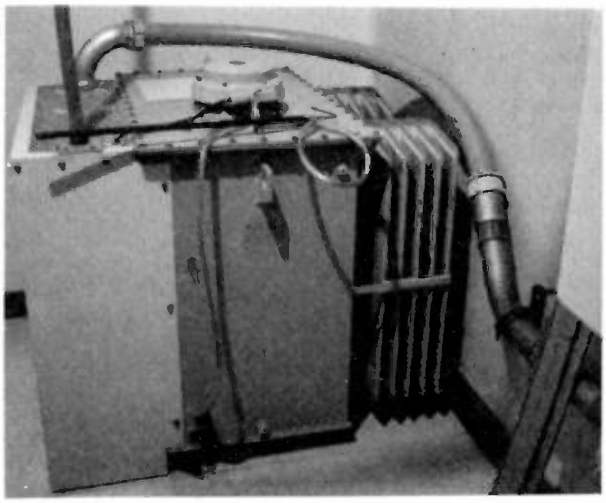
EPA REGULATIONS[5]

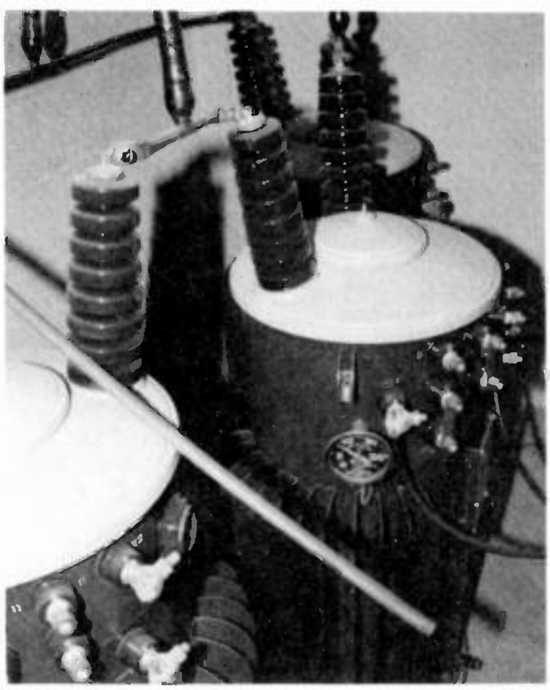
Under TSCA, the Environmental Protection Agency was charged with promulgating rules on PCBs which have subsequently been published in some 40 issues of the Federal Register. They are primarily set forth as "the law" in four issues, and cover the use, servicing, inspection, labeling, recordkeeping, storage, disposal and notification of others regarding possession of PCB Transformers and PCB spills. A full treatment of the regulations is beyond the scope of this article, but some of the important considerations will be capsuled here. For more detailed specifics, the reader is referred to the EPA rules as contained in 40CFR, Part 761.

These laws are being enforced and there are gravity-based penalties for non-compliance or illegal activities. Some owners have had to pay a high price financially for violations, and others an even stiffer penalty such as a public apology in a full page newspaper ad while serving a jail sentence. EPA takes the position that persons may not contract away their responsibility or liability for violation of the rules, so owners must be absolutely certain the firm they engage to do disposal has integrity, and assets to backup liability acceptance.

The EPA regulations divide electrical equipment into four major classifications for use and servicing, and two of these are of significant importance in the Broadcast Industry. Askarel transformers and Large PCB Capacitors have been widely used in transmitting stations, and askarel transformers may also be present in commercial buildings housing the studio or owned by the station.

A PCB Transformer is one which has 500 ppm (parts per million) or more of PCB in the fluid. For comparison, an askarel transformer will generally have 600,000 ppm or more.





Liquid-filled Transformers

The exposure risk to human food or animal feed from PCB Transformer use or storage is a possibility in some locations, and should have been eliminated by October 1, 1985. The user of a PCB Transformer posing food/feed exposure risk is responsible for weekly inspections, recordkeeping and maintenance until the user notifies the owner of such risk. Food/feed risk would not ordinarily be a factor in transmitting stations, but might be in studio buildings.

Any PCB Transformer requires quarterly inspections for leaks. If an impervious dike is added to contain all of the liquid material, or if the unit is retrofilled to be less than 60,000 ppm, the inspections may be yearly. If a leak is discovered, it must be contained, the inspections have to be daily and cleanup initiated as soon as possible, but in any case within 48 hours. Records have to be kept of the transformer existence, as well as details of the inspections and treatment of leaks, for 3 years after disposing of it.

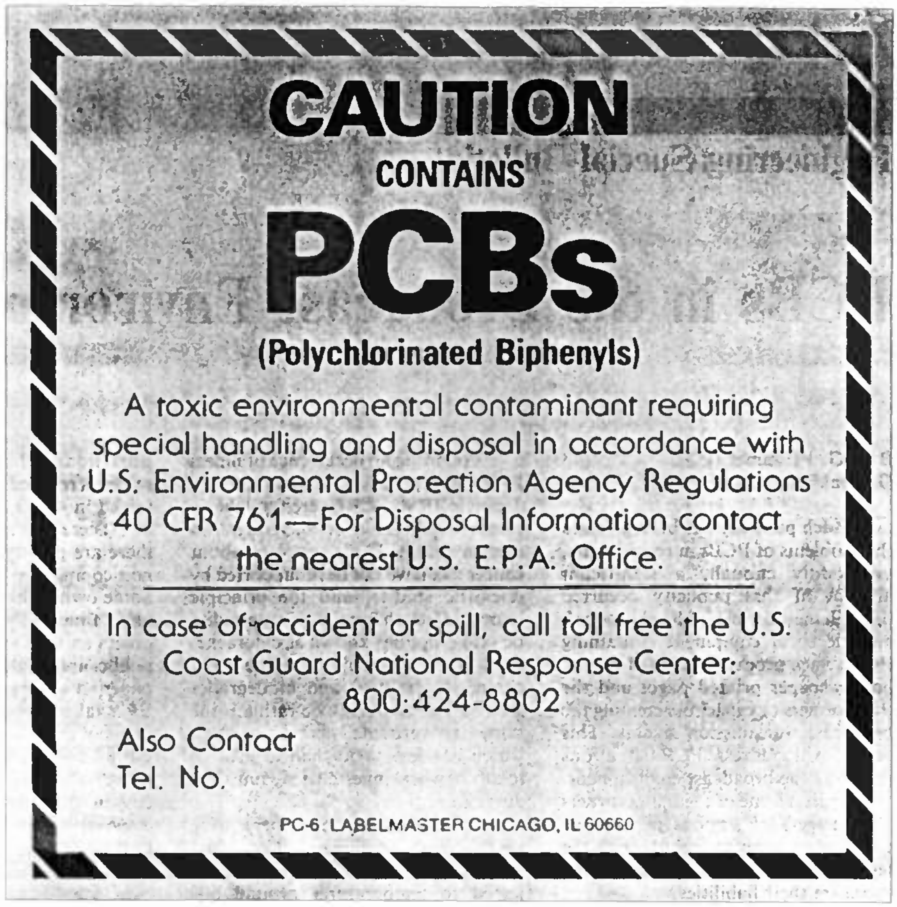
The PCB Transformer and doors or other means of access to it must have an EPA specified "Large PCB Mark" label applied, with particular requirements as to size, wording, border, and colors. Marking labels which meet the EPA-specified standards can be ordered from many sign/label vendors, including among others: (1) Seton Name Plate Corp. (203) 488-8059; (2) Brady Signmark™ Division (414) 961-2333; (3) Labelmaster 1-800-621-5808.

Combustible materials must not be stored within 5 meters of a PCB Transformer or its enclosure. The PCB Transformer has to be registered with local fire response personnel, and if used in or within 30 meters of a commercial building, the building owner must be notified. If the transformer is involved in a fire-related incident causing the release of PCBs, the owner must immediately report it to the National Response Center (1-800-424-8802) and take measures to contain and control releases into water.

A PCB Transformer which fails cannot be untanked for repair, and disposal of the liquid and carcass are strictly regulated by EPA to approved incinerators and chemical waste landfills. By EPA definition, a leak is considered illegal disposal.

Any PCB Article or PCB Container stored for disposal must be date-tagged and inspected for leaks every 30 days. It must be removed from storage and disposed of within one year from the date it was first placed into storage. Liquid PCBs above 500 ppm may only be stored in a special facility above the 100-year flood water elevation. Other PCB items may be stored with certain restrictions for up to 30 days in other than the special facility.

Samples of EPA-Specified Warning Labels



Disposal of all capacitors is regulated to some extent unless it is known that it does not contain PCB's. **Large PCB Capacitors** are those with three pounds or more of dielectric fluid, and size measurements can also be used to distinguish Large Capacitors. If not posing a food/feed risk, they may continue to be used until October 1, 1988, but after that only if the access is restricted and indoor installations will also have to contain any release of PCBs from the capacitors. Inspections are not required, but those operating at 2,000 volts or more must have the special Large PCB Mark label now and lower voltage units labeled when removed from service. Both high and low voltage Large PCB Capacitors must be disposed of by incineration.

Use, servicing and disposal of **PCB Small Capacitors** (less than 3 lbs of dielectric fluid) is not restricted by EPA unless there is a leak. In that event, the leak must be repaired, or the capacitor disposed of either by incineration or by placing in a specified container and burying in a chemical waste landfill.

Mineral oil-filled transformers are by EPA definition classified as **PCB-Contaminated Transformers** unless 500 ppm or greater fluid is known to have been added or they have been tested and found to be 500 ppm or greater, in which case it is a PCB Transformer. If tested by a specific gas chromatograph method and found to be under 50 ppm it may be classified **Non-PCB**. They do not have to be labeled or inspected, and can be untanked to repair a failure. Disposal of the liquid is regulated by EPA, and in some states both the carcass and the liquid are subject to additional regulations on transport and disposal, extending down to as low as 1 ppm.

Mineral Oil Transformer Nameplate

IDENTIFYING PCB ITEMS

Where are the PCBs in a Broadcast transmitting station? In the case of new equipment built subsequent to 1979, there probably aren't any, but it would still be advisable to check and know definitely what is actually present. A liquid-filled transformer will usually have some cooling fins on it, and the nameplate will also provide useful information on the type or contents. If the nameplate indicates any of the generic or trade names for askarel (see Askarel Table), or if there is no nameplate at all or the fluid is not identified, it must be classified as a "PCB Transformer". If it contains mineral oil (see Petroleum Oil Table), it is still classified as "PCB-Contaminated" until a test proves otherwise.

ASKAREL*	TRADE NAMES
ALC	R.C. Uptegraff
Apirolio	- - - -
Aroclor, Capacitor 21	Monsanto Co.
Asbestol	American Corp.
Ask	Queensboro Transf & Mach
Clophen	Bayer (Germany)
Clorextol	Allis Chalmers
Chlorinol	Sprague
Clorphen	Jard
Diaclor	Sangamo Electric
Dicanol	- - - -
DK, Inclor, Fenclor	Caffaro (Italy)
Dykanol	Cornell Dubilier
EEC-18	Power Zone Transformer
Elemex	McGraw Edison
Eucarel	Electric Utilities Corp.
Hyvol	Aerovox
Inerteen	Westinghouse Corp.
Inflamol	Italiano
Kanechlor	Mitsubishi (Japan)
Kennechlor	Kanegafuci Chemical Ind.
MCS 1489	Monsanto Co.
N-3	Niagara Transformer
Nepolin	- - - -
No-Flamol	Wagner Electric
Non-Flammable Liquid	ITE
Peneoclor	Prodelec (France)
Pydraul, Pyroclor	Monsanto Co.
Pyralene	Prodelec (France)
Pyranol, Magvar	General Electric Co.
Saf-T-Kuhl	Kuhlman Electric
Santotherm	Mitsubishi (Japan)
Santovac 1,2, Santhotherm FR	Monsanto Co.
Shibanol	Toshiba
Solvol	USSR
Therminol	Monsanto Co.

*Askarel is the generic name for PCB fluids.

Older liquid-filled units not in separately enclosed vaults are likely to be askarel, but if the nameplate leaves some doubt, it is advisable to consult the manufacturer or a knowledgeable expert on PCB and dielectric fluids.

PETROLEUM OILS	
Trade Name/Designation	Supplier
10C....................	General Electric Co.
10C-A (Inhibited)......	General Electric Co.
00600..................	Texaco
A13A3A1................	General Electric Co.
A13A3A2 (Inhibited)....	General Electric Co.
Caltran #60............	Calumet
Diala A................	Shell
FR Insul. Fluid (HTH)..	Gulf
Insulating Oil.........	Chevron
Metran 60, 60-15, 60-30.	Metalworking Lub. Co.
Mepsol (HTH)...........	McGraw Edison
R-TEMP (HTH)...........	RTE Corp.
Suntrans I.............	Sunoco
Suntrans II (Inhibited).	Sunoco
Transcrest H...........	Gulf
Transil................	Generic
Type I.................	NEMA
Type II (Inhibited).....	NEMA
Type 21 Transil Oil.....	1913 G.E. Transformer
Univolt 60.............	Exxon
WEMCO..................	Westinghouse
WEMCO FR (HTH)..........	Westinghouse

In older transmitters there will likely be several PCB Capacitors, both Large and Small. The rectifier cabinet will have several Large High-Voltage Capacitors and some may have leaks around the bushings or elsewhere, even though they have not failed. The leaked material may have spilled over the top edge and run down on the floor of the cabinet or underneath it. Such leaks must be repaired as soon as possible, or the unit replaced. The spill must be cleaned up and the resulting waste disposed of in an EPA-approved chemical waste landfill. Any old Large Capacitors which are removed must be incinerated.

A Leaking Large PCB Capacitor

Spare Large PCB Capacitors

The Large High-Voltage Capacitors in use, as well as spares, must have the Large Mark label now, and a record of their existence and final disposal entered in the inventory log book. They do not have to be inspected for leaks, but bear in mind the October 1, 1988 deadline on containment and access restriction.

The rectifier, exciter/modulator and power amplifier cabinets also contain a large number of Small Capacitors, which in older equipment are probably askarel-filled. They pose no particular problem unless leaking or exploded, in which case disposal is then restricted to containerized chemical waste landfill burial or incineration, and the spilled material must be cleaned up and disposed of as PCB solid waste.

The liquid-cooled rectifiers in the rectifier cabinet may contain askarel, and if so, even though their use is not regulated, disposal as a PCB Article with 50 ppm or more PCB is regulated to incineration, or a chemical waste landfill if first drained of free-flowing liquid which is then incinerated.

Leaked PCB's On Cabinet Floor

Small PCB Capacitors in Rectifier Cabinet

Small PCB Capacitors in Power Amplifier Cabinet (upper right)

MANAGING PCBs

Improper disposal because of not knowing the transformers or other items contain PCBs can be very costly. In the case of one radio station which was demolished following a fire, the debris hauled to a County landfill recycling center unknowingly included some small PCB Transformers. An insulator bushing broke on one when it was dumped out, and subsequently some 70 tons of garbage became PCB contaminated. The recycling pit had to be scrubbed clean, and three truckloads of garbage transported to and buried in a chemical waste landfill instead of the County landfill, all at a cost of several hundred thousand dollars.

In another instance a broadcasting company donated transmitting station property and buildings to a philanthropic cause, only to have it come back to haunt them later when discarded PCB items were found on the premises by the new tenants. Large amounts of soil excavation and disposal were then necessary at considerable cost.

A radio station Chief Engineer contacted a locally advertised hazardous waste emergency response company to ask about disposal of a small PCB capacitor which had exploded. He was told that since it was under 3 pounds, he could wrap it up and throw in the garbage can. That would have been illegal disposal.

Under some circumstances, "used" PCB Transformers and Capacitors can still be resold in the U.S., if they were bought for use rather than resale. However, it is **illegal to export** PCB items without an EPA exemption. So if a PCB-equipped transmitter is to be sold to a broker or a buyer in another country, the owner's legal counsel and/or the Region EPA Administrative Office should be contacted.

Before an EPA visit occurs, obtain the services of a PCB-knowledgeable and qualified vendor to assist in making sure that all the regulations have been complied with. The most frequent violations found reportedly are improper or inadequate labeling, inspections not being made, records not being kept, and equipment leaks. **Citations and fines in such situations can be severe.**

There are many other regulations applying to storage, transporting, testing, use of PCB Transformers in commercial buildings and the like which are too involved for treatment here, but which should also be investigated and implemented where applicable.

For additional information and assistance you may wish to consult a local GE Apparatus & Engineering Services representative, who can be contacted through phone no. 1-800-626-2001, Ext. 722.

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1. "PCB Perspectives", National Electrical Manufacturers Association

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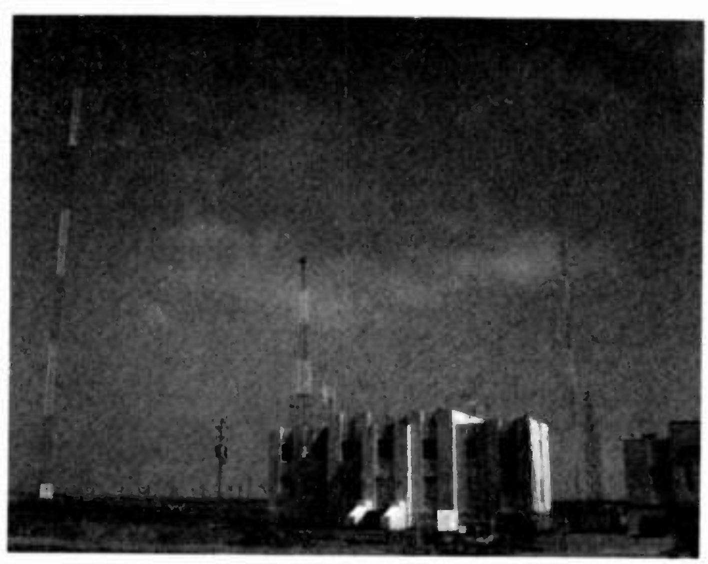
3. Clinical Toxicology of Commercial Products, Williams and Wilkins Company, Baltimore, Maryland.

4. "Causal Agents of Yusho", American Journal of Industrial Medicine, 5: 45-58 (1984).

5. 40CFR, Code of Federal Regulations, Part 761



National Association of
NAB
BROADCASTERS

FURTHER DEVELOPMENTS IN AURAL STUDIO TRANSMITTER LINKS

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Barry Victor
The Victor Group
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Abstract: S.T.L. Space in many markets is becoming increasingly scarce and unavailable. This report delineates the results of a series of efforts on the part of the Southern California Frequency Coordinating Committee to further exploit the limited spectrum currently available in the 950 MHz. and other bands.

S.T.L. Spectrum in major markets is still congested to the point that no additional stations can be accommodated. The plan the Southern California Frequency Coordinating Committee has to utilize the space between the existing channels is still being investigated and is very close to actual inplementation.

Exhibit A reveals the clutter and congestion in the Los Angeles 950 Band.

In the paper presented last year, we proposed a possible solution to the congestion. This included the use of 25 kHz. deviation on newly implemented narrow channels, and the use of packet radio techniques to remotely control transmitters.

A number of possibilities arose in terms of using these narrow channels; the original plan advocated last year called to conform with the FCC's 25 kHz. channel block plan (which recommended a maximum channel of 300 kHz. for FM and 200 kHz. for AM operators). To implement the plan without detriment to existing S.T.L. users, we tried using two monaural channels between the existing 500 kHz. apart channels already occupied by wide-band systems. These were 200 kHz. above the lower channel and 200 kHz. below the upper channel. Using both 25 kHz. and 50 kHz. deviation, our tests revealed the probability of success if the existing wideband stations used equipment with adequate adjacent channel rejection. Such receivers are available from a number of sources.

During our Los Angeles tests, in only one instance did a carrier 200 Khz. away from an in-use wide band system cause problems, and the manufacturer of the affected receiver has indicated a circuit correction should solve the problem.

The FCC's proposed band scheme is split up into 25 kHz. blocks, with the actual band coordination falling on the shoulders of local Frequency Coordinating Committees.

Even though the Commission feels the 200-300 kHz. plan will be adequate, our need to allow for anomalies or frequency drift in equipment, to maintain adequate guard bands and to use subcarriers leads us to prefer a more liberal and protected 250-350 kHz. plan.

Last year we elaborated on a system using two relatively narrow band (+/- 25 kHz.) systems, operating between wide band channels. We explained a series of tests that had been conducted, and were optimistic about the results. We did not consider the possibility of using composite stereo transmission in conjunction with narrow deviation.

This year we have the results of some experiments conducted between Hollywood and Mt. Wilson, using a narrow deviation composite system using subcarriers.

These tests were to validate some very encouraging results that had been obtained in a laboratory.

The carrier frequency of this system was at 942.7 MHz, placing it just 300 kHz. below the S.T.L. frequency of station KFAC, at 943.0 MHz.

Test Procedure

The stereo tests were performed using 35 kHz. deviation and 300 kHz. allowed bandwidth. The results indicated S.N.R. versus R.F. signal level, distortion, crosstalk, and separation. Monaural S.N.R. and T.H.D. measurements were also made on the 67 kHz. and 110 kHz. subcarriers.

Prior to the field analysis, factory control tests were run to gather base line data. These results would be compared to the field test data to spot any performance degradation caused by adjacent channel interference and/or other field conditions. The two factory control tests in the comparison were:

1) Back-to-back system performance without an adjacent carrier.

2) Back-to-back stereo S.N.R. at R.F. input levels of 30 uv., 100 uv., 300 uv., 1 mv., and 3 mv. with one interfering adjacent carrier 300 kHz. removed at a fixed R.F. level of 1 mv. This carrier was modulated +/- 50 kHz. with a 2 kHz. L-R composite signal.

The above tests were then re-run between the dual KTTV facilities at Hollywood and Mt. Wilson with the following exceptions:

1) Due to transmission line losses the maximum desired receive signal level was 300 uv.

2) The received level of the KFAC adjacent channel composite signal was 23 uv.

3) Another adjacent mono signal was added at 942.5 MHz. (200 kHz. below the test system) at a received signal level of 30 uv. to test its effect on S.N.R.. (KIIS-FM, in Los Angeles, holds a grandfathered license to operate at 942.5 MHz, but currently is inactive on the channel).

Results of the tests are included as Exhibit B.

Despite the encouraging results of the narrow band composite system tests, the occupied 300 kHz. bandwidth won't fit into the existing spaces in the Los Angeles S.T.L. spectrum.

What will fit into the existing spectrum are the narrow band monophonic systems we reviewed and spoke of last year. These use very narrow occupied bandwidths, and two of them will fit between current wideband systems without detriment. Exhibit C shows the Proposed Band Plan for this method of operation.

After testing was completed on the narrow band composite system, it was decided to simulate monaural conditions. To do this, the composite output of the receiver was disconnected from the stereo demodulator and fed to a deemphasis network to drive an audio voltmeter and spectrum analyzer for S.N.R. and T.H.D. measurements. The output of the transmitter was attenuated 20 db. to give a receive signal level of 30 uv. for the desired signal. The levels of the test transmitter 200 kHz. below and KFAC at 300 kHz. above were at 30 uv., and 23 uv. respectively. The deviation on the test transmitter was backed off 3 db. for +/- 25 kHz. deviation. Under these conditions, the test results were:

7.5 kHz. T.H.D. = 0.03 %
Uncorrected S.N.R. = 72 db.

Because of the difference in baseband bandwidth between a composite (80 kHz.) and a mono (15 kHz.) radio, the estimated correct monaural S.N.R. would be:

Corrected S.N.R. =
72 db.+ 10 log (80/15) = 79 db.

[See Exhibit D]

It's obvious excellent performance is available in a narrow band monophonic radio.

A number of manufacturers are working on advancing narrow band technology, and we expect to have additional results available after this goes to press.

It's likely that further away in time the technology possible with both monophonic and composite narrow band systems will necessitate a new band plan in congested markets. This should be decided on a market-by-market basis, and could involve moving center channels of existing users to accommodate various options. Among these are:

a) Using a narrow band composite system between current wide band operators.

b) Adding two new monophonic cross-polarized systems in the same space.

c) Replace existing wideband systems with narrowband equipment to accommodate more users.

The expenses of changing carrier frequencies to accommodate the new systems would be borne by the new licensee.

Other S.T.L. technology is being considered for use. As this is being written a digital 23 GHz. radio had still not been delivered for field tests. The use of digital transmission techniques are expected to improve the usable path length. It is hoped that a usable path of 10 miles will be possible while maintaining an adequate fade margin.

Many radio broadcasters can look upon all these developments with detached interest; their markets have no congestion and no need to address squeezing more into the available S.T.L. spectrum. Even so, they would do well to look to the future as new systems are installed. By addressing the new technologies congested markets have commanded they may save themselves anguish and expense by planning for the future. After all, 20 years ago even Los Angeles didn't have a congestion problem, phone lines were cheap, and certainly none of us anticipated the situation we are in now.

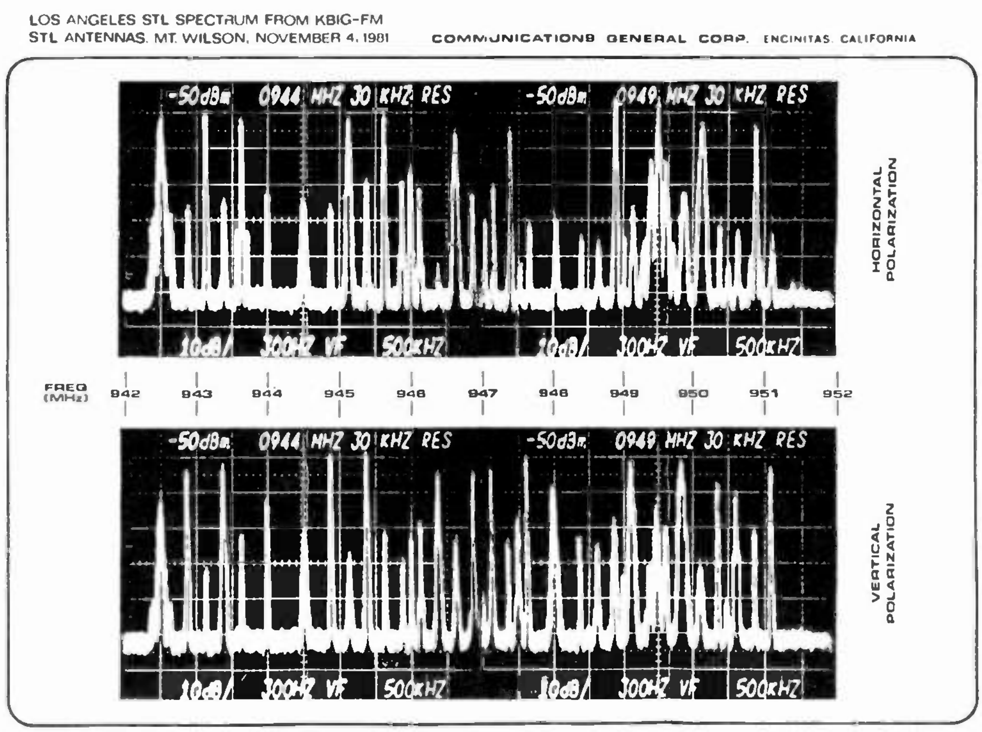


Exhibit 'A'

Test Results

	Stereo S.N.R. @ Input Level Of				
	3 mv.	1 mv.	300 uv.	100 uv.	30 uv.
Factory Control No Interference	78 db	74 db	65 db	55 db	45 db
Factory Control w/ Interference	78 db	73 db	64 db	54 db	-
Mount Wilson w/ KFAC	-	-	63 db	54 db	44 db
Mount Wilson w/ KFAC and PCL-303 @ 942.5	-	-	63 db	54 db	44 db

		Factory Control No Interference	Mount Wilson With KFAC
Stereo Distortion @	460 Hz.	0.07 %	0.08 %
	7 khz.	0.07 %	0.08 %
	15 kHz.	0.24 %	0.30 %
Non-Linear Crosstalk	Main to Sub	62 db	60 db
	Sub to Main	70 db	65 db
Separation @	40 Hz.	50 db	44 db
	400 Hz.	49 db	44 db
	15 kHz.	52 db	50 db
67 kHz. Subcarrier	S.N.R. @ 300 uv.	54 db	53 db
	T.H.D. @ 400 Hz.	0.5 %	0.6 %
110 kHz. Subcarrier	S.N.R. @ 300 uv.	43 db	42 db
	T.H.D. @ 400 Hz.	0.8 %	0.9 %

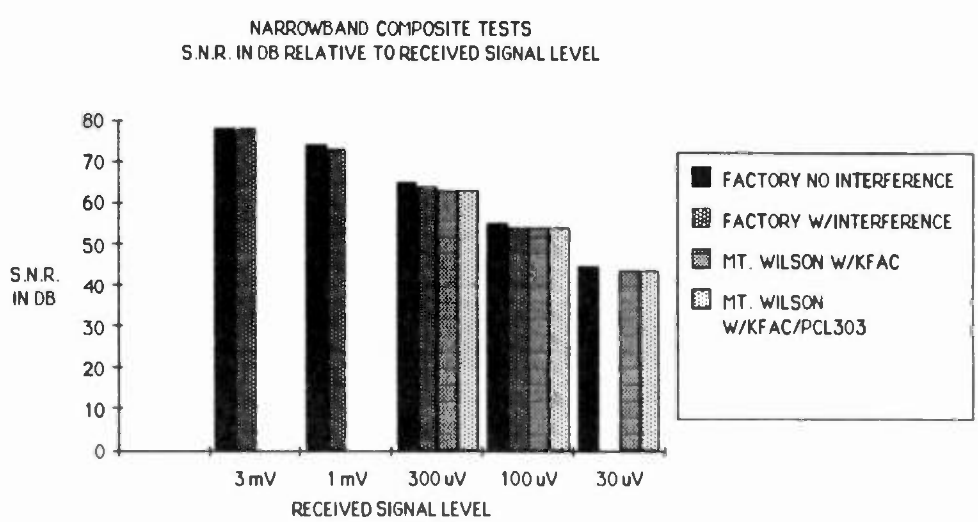


Exhibit 'B'

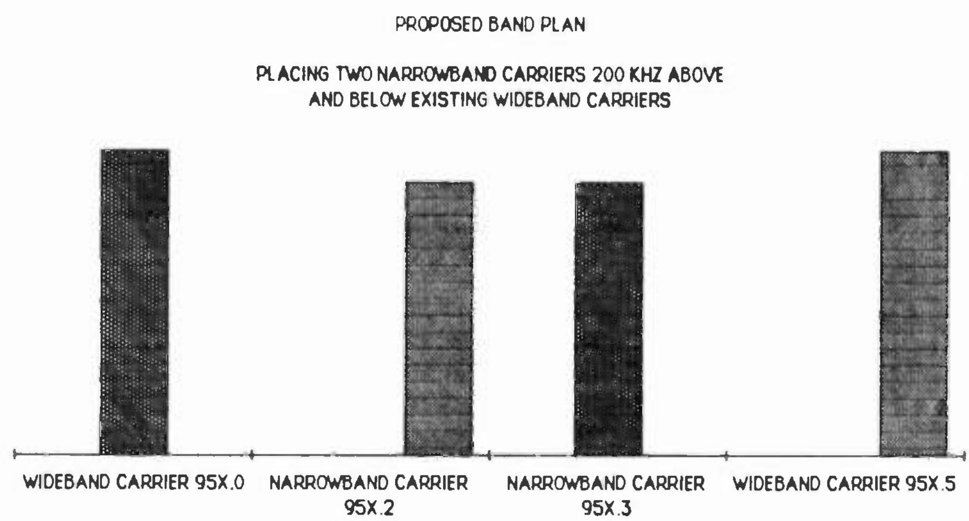
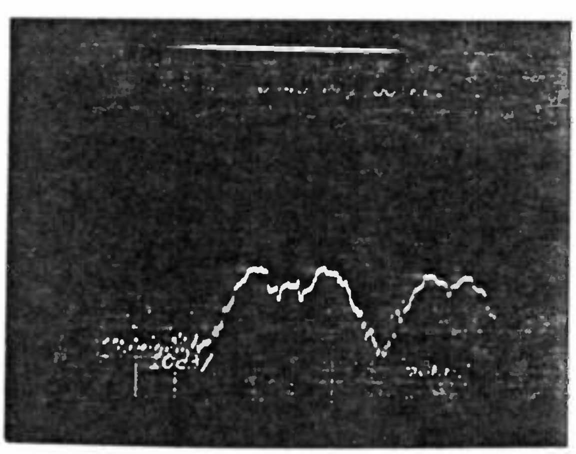


Exhibit 'C'



Horizontal: 10 dB/div

Ref. Level: -30 dBm top scale

Vertical: 50 kHz/div

Res. BW: 30 kHz

Exhibit 'D'

Occupied bandwidth of mono narrowband test.
Center carrier is wideband PCL-303, at 942.5 mHz.
Right carrier is narrowband mono carrier, at 942.7 mHz.

INSTALLING AND OPERATING A 23 GHZ RADIO STL SYSTEM©

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ABSTRACT

This paper will introduce the radio broadcast engineer to an alternative Studio to Transmitter Link system for use in short microwave jumps found in large metropolitan areas.

In this paper, I do not plan to delve into the theoretical operation of this system, but plan to take a "real world" approach to the installation and operation of this system

This paper will focus on the MA/COM 23 VFM video microwave system and the dbx 700 Digital Audio Processor. Although there are other FM video microwave systems, as well as other types of digital audio processors, the author will deal with the (MA/COM)/dbx system because it is what is in operation at the authors station.

INTRODUCTION

Since the beginning of split site broadcasting, broadcasters have been in need of ways to get the program material from the studio to the transmitter site. Equalized phone lines were first used to send the program material to the transmitter.
In some areas, the quality and dependability of these lines leaves a lot to be desired. In many cases it is not the quality of the lines, but cost. Since the breakup of A.T.&T., many stations found their program loop costs skyrocketing. So many broadcasters have decided to go with an S.T.L. or Studio to Transmitter Link system. Authors note: Unless otherwise noted, "transmit" will note the studio end of the STL system and "receive" will note the end at the transmitter site.

The first Studio to Transmitter Link system was a simple transmitter/receiver setup. The program audio was placed into the audio input of the transmitter and then sent over carrier to the receive site, where the receiver returned the signal to audio. The audio was then sent to the transmitter via any processing. When broadcasters began to broadcast in stereo, two such units were used, one for each channel. When the 950 MHz band was opened to broadcasters, the composite STL system was developed. On the studio end, the Left and Right program audio are combined into a composite signal. This composite signal is actually the stereo signal that will be broadcasted. The composite signal containes (L+R, L-R, and the 19 kHz pilot.) On the receive end, this composite signal can either be demodulated back into the descrete left and right, or left as is and fed directly into the wideband input of the FM exciter. The major problem with the 950 MHz band today is crowding. In many of the large markets, finding an empty 950 MHz frequency is simply impossible. This problem is especially prevalent where several broadcasters are broadcasting from a common point, such as Sears Tower in Chicago, or the Prudential Building in Boston.

The system about to be described is very suitable for short distance microwave links and consists of two basic components. The R.F. components consists of an FM wideband video (yes I said VIDEO) microwave system located in the 23 GHz (23,000 MHz) band. Although this band of frequencies is not allocated to the broadcasting industry, it is located in the comerical fixed microwave station service, we still can use this band of frequencies. The microwave system that will be described can be ordered with one or two audio subcarriers. But we will not use these subcarriers to send our program audio. We will instead use a second piece of equipment. Enter the dbx 700 Digital Audio Processor. The dbx 700, in its simplest form, takes the program audio, left and right, and will turn them into digital video. This "video" will then be sent over the video microwave to the receive site. At the receive site the video will be decoded and returned back to left and right audio, where it will be sent to whatever processing and then to the broadcast transmitter.

R.F.

The R.F. part of this system uses a wide-band FM video microwave system, such as the MA/COM MA-23VFM. This system is ideally suitable for intracity microwave links. There are other systems capable of longer shots, but for the purposes of this paper, we will keep our focus on the short range system.

The MA/COM MA-23VFM system is capable of transmitting a full 8 MHz bandwidth without any audio subcarriers (for high resolution video), as well as a full color picture with two audio subcarriers. The MA/COM comes standard with one subcarrier, with the second one as an option. In the standard configuration, the video takes up the first 4.2 MHz of the bandwidth, with the first audio subcarrier located at 6.8 MHz and the second audio subcarrier located at 6.2 MHz. The audio channels are each 15 kHz wide and capable of transmitting data at 9600 baud, with use of an external modem. Something to keep in mind, many remote control systems operate at 1200 baud. The subcarrier, in use with a Telemetry Return Link (TRL) system can give the engineer the ablilty to put the remote control system on the microwave link. Stations that broadcast an SCA could also use this extra channel to send the SCA information to the transmitter site.

Each side of the MA/COM system has two components. A control unit, and the actual R.F. dish assembly. On the transmit side, the control unit takes the video and the audio subcarriers and combines them into a composite baseband signal at 70 MHz. This 70 MHz baseband is then sent to the RF/dish assembly, along with the D.C./control on a seperate cable. This baseband then frequency modulates (FM) a 23 GHz Gunn oscillator. The system has a deviation of 4 MHz. On the receive end, the 23 GHz is down converted to a 140 MHz I.F. The I.F. is then sent to the control unit where is undergoes a second down conversion. This signal is then sent to the baseband demodulator where the video and audio are seperated. The video is sent through a video filter and the audio is sent to seperate subcarrier demodulators, one seperate demodulator for each audio subcarrier. (Fig. 1)

AUDIO

As previously mentioned, the MA/COM can be ordered with two 15 kHz audio subcarriers. So, why not use these subcarriers to send our program audio to the transmitter? There is a way to generate a very high quality audio using a second piece of equipment. How high quality? 105 dB s/n with +/- .05 dB variation from 20 Hz to

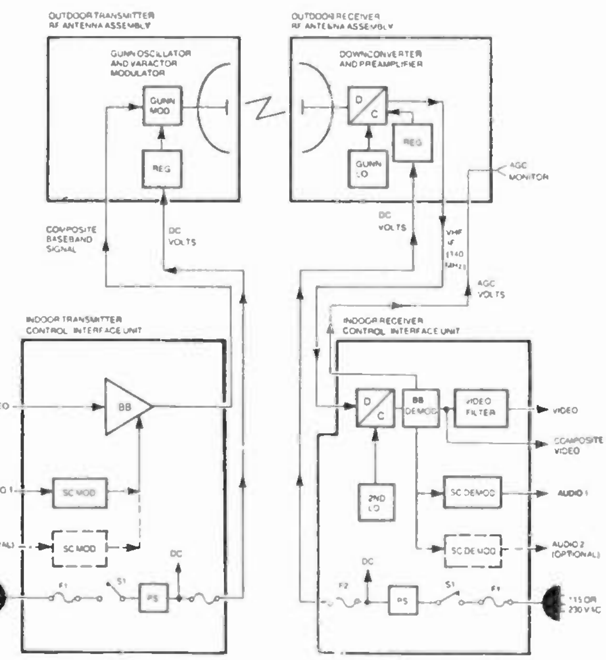

Fig. 1

20 kHz. By use of the second piece of equipment, the dbx 700, we can now send a high quality audio to the transmit site.

The dbx 700 Digital Audio Processor is an analog to digital convertor or encoder, on the transmit side. On the receive side, the process is reversed. The dbx 700 does not work by way of a 16 bit A to D convertor, but by way of a "Companded Predicitive Delta Modulator"(TM) or delta modulator for short. In Pulse Coded Modulation (PCM), the amplitude levels are set up as quantized levels, i.e. 1 volt = binary code 0001, 2 volts = 0010, 3 volts = 0011 and so on. Then we look at the input signal in relation to these levels, as well as to time.[1] (fig. 2)

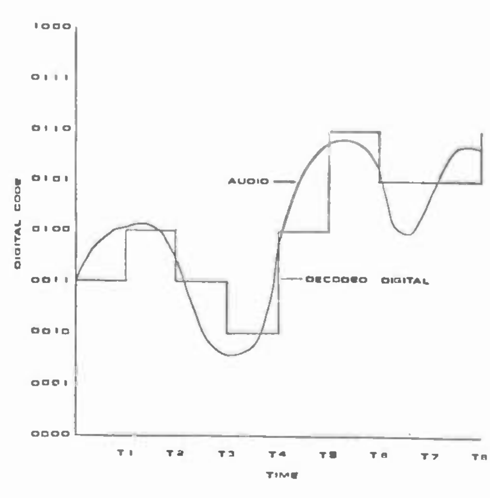

Fig. 2

In delta modulation, sometimes called "slope modulation", the encoder does not record the amplitude of the audio signal, but the change in the signal. The encoder only transmits its information as to whether the analog signal it is encoding is to "go up", or "go down".[2] (fig. 3) The rate at which the change in the signal is recorded is called a sample rate. In the case of the dbx 700, the sample rate is 644 kHz. (fig. 4)

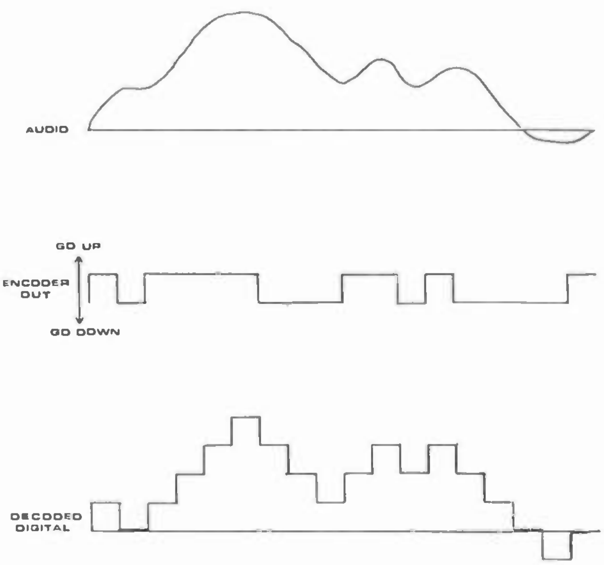

Fig. 3

Each digital encode/decode scheme has its own advantages. PCM reacts well with a rapidly changing signal. Delta modulation is much less sensitive to bit errors. Unlike 16 bit PCM, where a MSB (Most Significant Bit) error will cause a large output spike, delta modulation has no MSB. Random burst errors as large as 30 bits are frequently inaudible during program material.[3]

The companding process in "Companded Predictive Delta Modulation"(TM) is simply Compressing before digital encoding, and Expanding after decoding.

That's the basic overview of the system. Now let's put all the pieces into play.

PATH ANALYSIS

As with any STL system, the first step is to plot your STL path and preform a path analysis. 7.5 Minute Topigraphic Quadrangle maps should be used. These are the same maps use for plotting radials for AM Dirctional Antenna systems and can be optained from the United States Geoligical Survey.* Maps will be needed for the areas that the STL will be going through. Once the path has been plotted, the next step is to analyze it.

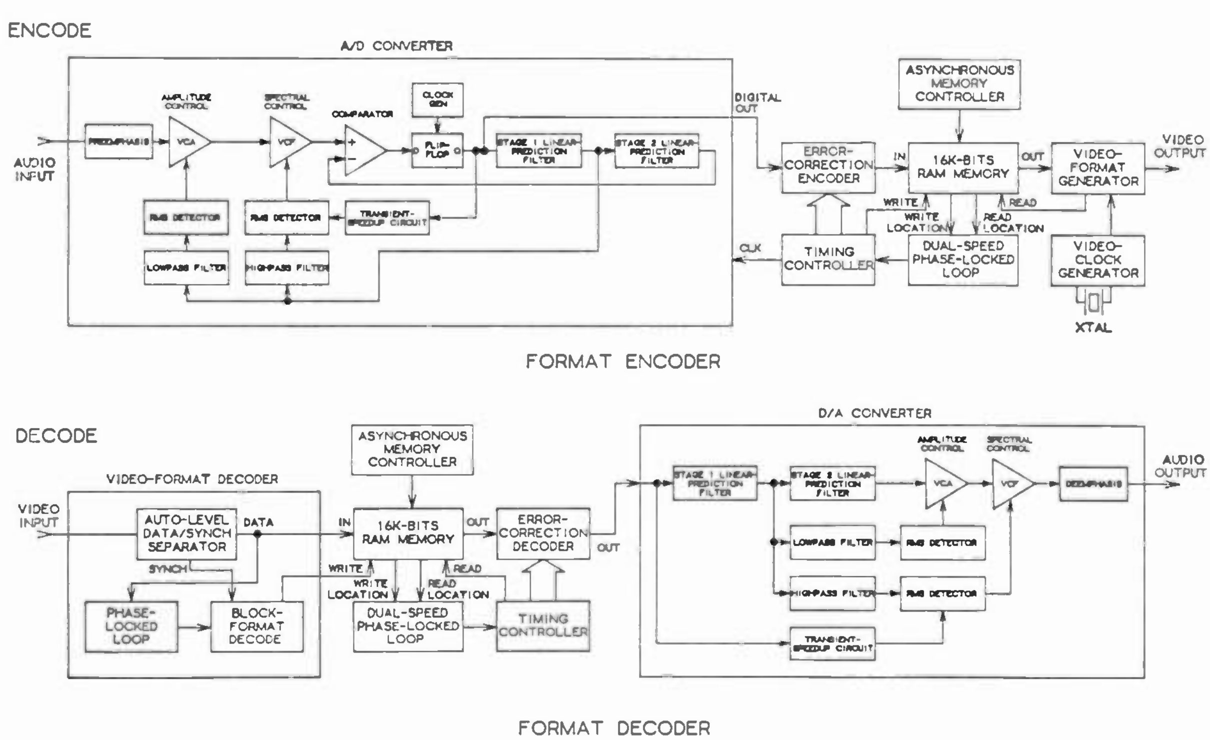

Fig. 4

<u>Given</u>

The following are the given conditions for the path and equipment at WMJX FM in Boston:

Path Distance	.50 Miles
Antenna Gain:	
Transmit	33 dBi
Receive	33 dBi

Transmitter output power:

Minimum	17 dBm	(50 mW)
Maximum	20 dBm	(100 mW)
Typical	18 dBm	(66 mW)

<u>Path Loss</u>

The path loss for 23 GHz is given by the following equation:

$$L_p\ (dB)=+G1-36.6-20\log(f)-20\log(D)+G2 \quad (1)$$ [4]

where G1 is the gain of the transmit dish, G2 is the gain of the receive dish, f is the frequency in <u>MHz not GHz</u>, and D is the distance of the path in miles. Substituting the givens for the WMJX STL path, results in a calculated path loss of 51.9 db. The transmitter puts out 18 dBm of power, we'll assume the typical case, so the expected signal at the receiver is given by:

$$(\text{Transmitter output power}) + (L_p) \quad (2)$$

Once again substituting gives an expected signal strength of -39.1 dB at the receiver. The receiver will function at -72 dB, so the equation for fade margin is simply Minimum receive level - expected signal strength. In the case of the WMJX path, the fade margin is 32.9 dB. It is recommended that a fade margin of at least 30 dB is maintained. Systems that have smaller fade margins may become more susceptible to outages.

LICENSING

Once the path has been analyzed, the next step in the process is to obtain a Microwave Radio Station License. Before the F.C.C. will accept any applications, an interference study must be completed by a recognized database, such as Spectrum Planning. In addition to doing the required survey, the company doing the survey will also assist the applicant in the proper paperwork to fill out, as well as protect the applicant's frequency during and after the application has been filed. The application for the license is filed, not on F.C.C for 313, but on form 402. At the present time, there are eight (8) 50 MHz channels:

21.825 GHz
21.875 GHz
21.925 GHz
21.975 GHz
23.025 GHz
23.075 GHz
23.125 GHz
23.175 GHz

INSTALLATION

Now it's time to install the microwave system. We'll assume that you have done your path analysis, applied for and have received your Microwave Radio Station License and are now ready to install your microwave system. The first thing to do is to decide where you are going to locate the control unit and the RF/Dish assembly. The control unit and the RF/Dish assembly must be placed within 250 feet of each other. Once the location of both units has been established, it is then time to deal with the control units.

The MA/COM control unit is simply a box with a switch for controlling the transmiter or receiver. It would have been nice if MA/COM had built this unit into a standard 19" EIA rack mount. No problem, instant rackmount. Taking a 5 1/4" rack panel, the control unit was centered on it. A hole was cut that would allow the front panel of the control unit to stick out from the panel, and the unit was held in place by two 90^{o} angle brackets. The front of the rack panel was then painted flat black to match the dbx 700. (fig. 5)

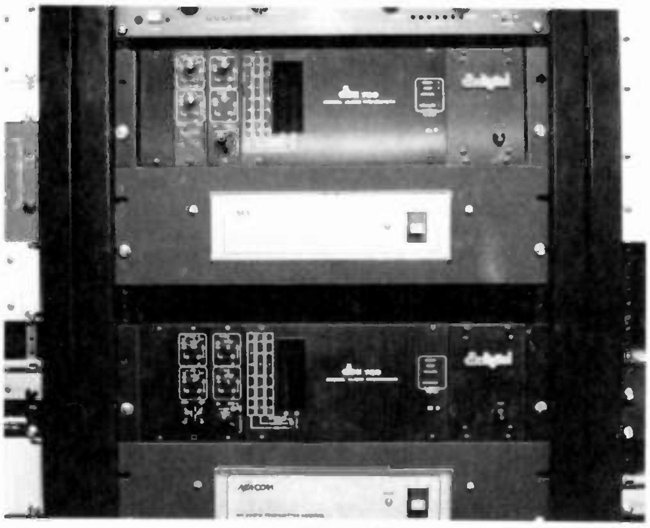

Fig. 5

The RF/Dish assembly mounts to a tower or pipe just like any conventional side mounted dish or antenna. Now that the control unit and antenna assembly are mounted, the next step is to connect them. (fig. 6)

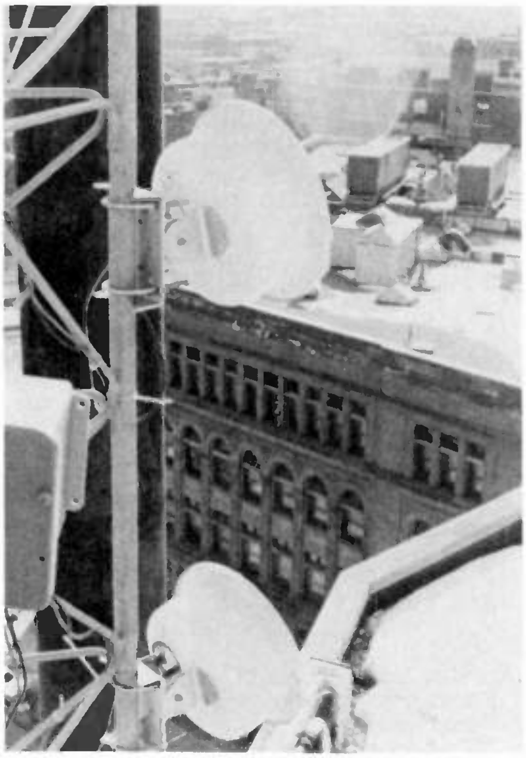

Fig. 6

MA/COM recommends that RG-6/U coaxial cable be used for the baseband video between the control unit and the RF/Dish assembly, and that a 4 conductor # 20 AGW shielded cable be used for the D.C.

Transmit

The studio is located in Back Bay Boston on the second floor of an 11 story building. On the top of the building, the station has a 50 foot tower. On this tower, the station maintains 5 STL antannas. Three of the dishes are 950 MHz, while the other two are the 23 GHz MA/COM RF/Dish assemblies. (fig. 7) The distance between the control units and RF/Dish assemblies is approximately 200 feet. On this side, the recommended cables worked fine. Going from the output of the dbx 700 to the input of the MA/COM control unit, a 10" piece of RG-58 coaxial cable was used. The receive site was a totally different story. (fig. 8)

Receive

The WMJX transmitter plant is located on the top (literally) of the Prudential Building in Back Bay Boston with the RF/Dish assemblies mounted to a guard rail outside the transmitter building. (fig. 9) After an attempt was made using the recommended cables, a more . . . drastic step was taken. The RG-6/U was replaced with 1/2" heliax, and the

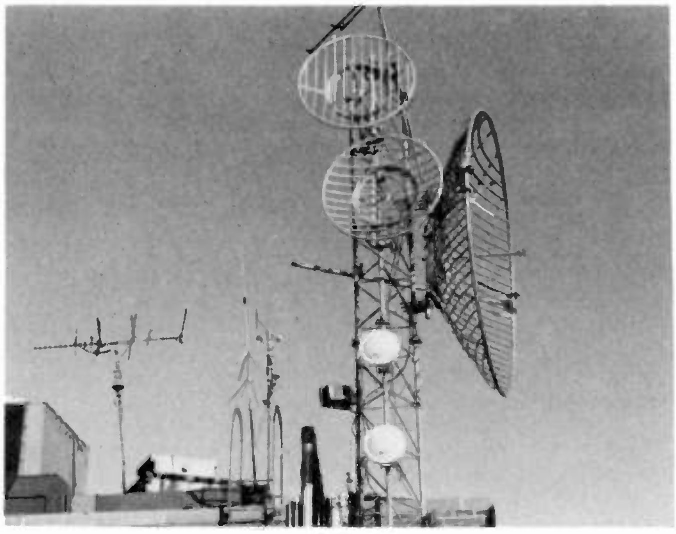

Fig. 7

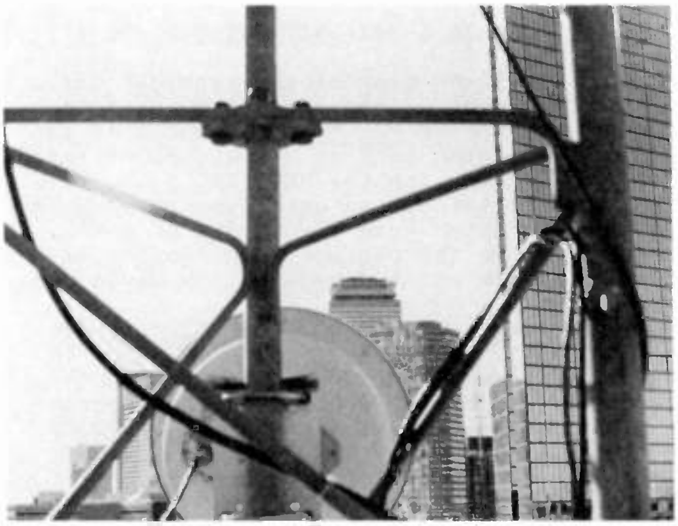

Fig. 8

RG-58 cable that went between the dbx 700 and MA/COM control unit was replaced with 1/4" heliax. The MA/COM control unit was modified by replacing the I.F. input BNC connector with a type N connector. Ferrite beads were placed on all leads that came from the back panel, and the internal video cable, the cable from the circuit board to the BNC connector on the back, was replaced with RG-174 coaxial cable. (fig. 10) The dbx 700 was placed in a shield of tin foil. (fig. 11) An R.F. tight box is currently on the drawing board.

Is all this really necessary? For the vast majority of stations, the system will work straight out of the boxes. To understand why it was necessary to go to the extreme measures we had to, let's take a look at the roof of the Prudential Building.

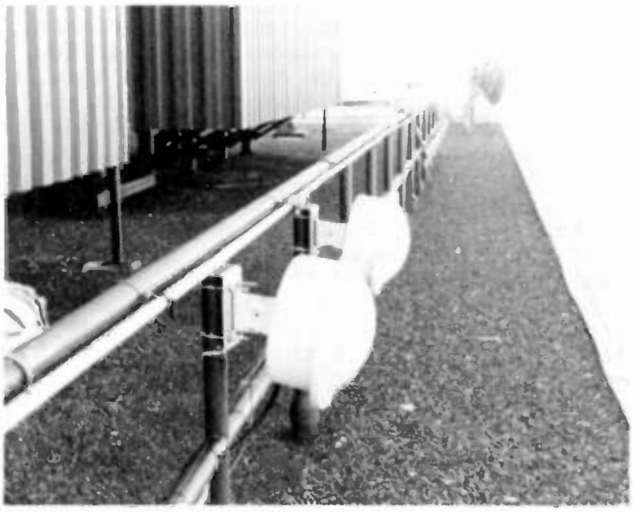

Fig. 9

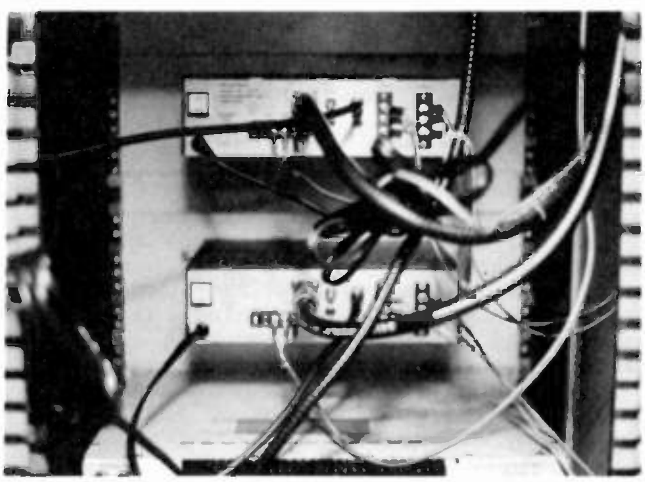

Fig. 10

Four commerical broadcast stations at 25 kW each, one noncommerical broadcast station with 3 kW, one full power UHF television station (channel 68) at 1.35 MW E.R.P., and several two way systems. In addition to the above, there are several microwave stations as well as three E.N.G. receive sites on the roof. To top that off, a low power television station will soon begin operation. It is not that there is a lot of stations operating on the roof, but that they are operating in close proximity to each other. The average voltage "E" density arround the WMJX transmitter building is approximately 150 V/M. The average E field inside the transmitter building is 15 V/M and the average E field inside the rach housing the dbx/MA/COM system is 10 V/M. (fig. 12)

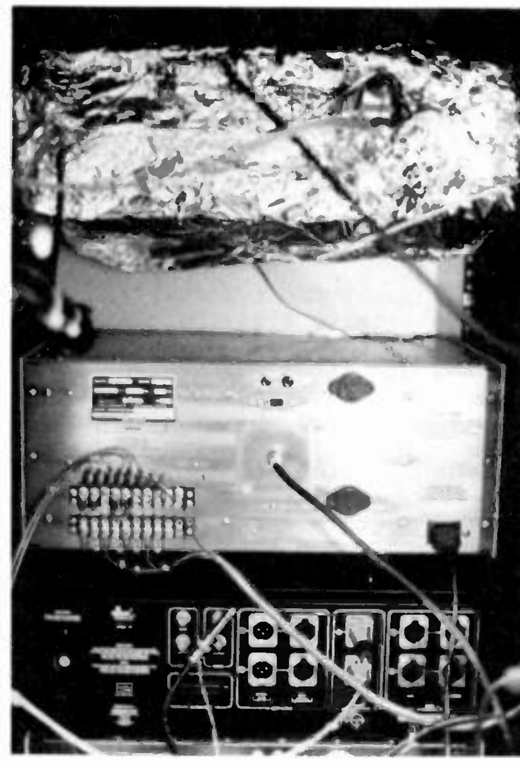

Fig. 11

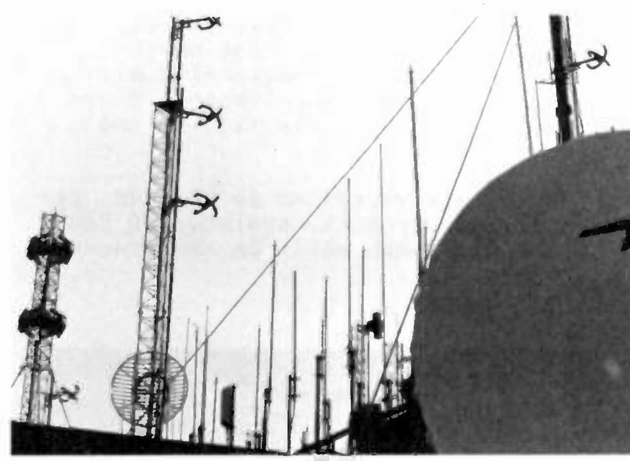

Fig. 12

ALIGNMENT

Now that the microwave system has been installed, it is now time to align it. The MA/COM manual recommends the following procedure:

Turn the system on and point the transmit antenna is the direction of the receive antenna. Using the plug at the back of the dish, which is a remoted AGC voltage, Plug a D.C. volt meter into the socket, and peak the AGC voltage. First by sweeping the dish left and right, then up and down. Once the maximum AGC level has been obtained, repeat the procedure with the transmit dish, using a telephone or two way system for communications. This procedure works well, but due to the R.F. on the roof of the Prudential Building, an alternative method was utilized.

Once again, point the transmit antenna towards the receive site, but this time put video, not the dbx, on the system. A color bar generator works great if you have one. If not, a color video camera pointed at an object will work as well. On the receive site, hook up a color monitor to the video output of the MA/COM. Now, sweep the dish, looking for the minimum video noise possible. Remember, as you sweep through the signal, you will go through three points that the signal will exists. Lock on to the strongest one. Locking on to a minor lobe of the transmitter will reduce the signal by 20 dB! So much for the 30 dB fade margin. Repeat the process with the transmit dish, once again using two way radios or telephone for communications. The reason for using a video signal to align is simple. It is much easier looking for something that is familiar, i.e. a picture or color bars. In addition, if you suspect that the link is being interfered with from another source, tracking is simplified. Most of the users on this band of frequencies are using it for video transmission. There is a better chance of identifying the source in question.

Once the microwave system is aligned, remove the video from the system, and put the dbx 700 on each end. On the transmit side, apply program audio to the dbx 700 and set the input module switch to record. Figure 13 is what the waveform out of the dbx 700 looks like. On the receive side, take the video output from the MA/COM and apply it to the input of the dbx. Place the input module switch to the play position. If the link has been set up correctly, the video unlock light will go out and the stand by light will come on. After two or three seconds the stand by light will go off and the video lock lamp will be on. Adjust the slice level for minimum error correct, that is, adjust the slice level until the error correct goes out, or until it only blinks occasionally. It is normal for the error correct to blink occasionally.

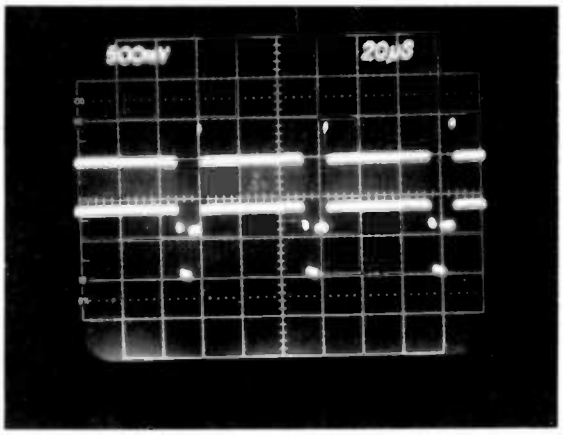

Fig. 13

Once the system has been set up, it should seldom require adjustment. Your 23 GHz STL system is now ready for use.

OPERATION

The 23 GHz STL system operates just like any other STL system, with a few differences.

Rainout

23 GHz is not like 950 MHz in the sense that 950 is not affectied by rain to the degree that 23 GHz is. There is a reason maintaining a 30 dB+ fade margin. When it rains, water builds up on the radomes of the antennas. The buildup can antenuate the signal as much as 5 dB per antnna. In addition to the water buildup on the radomes, the falling rain can anteuate the signal. During a sudden downpoor or cloud burst, the signal may be antenuated by a much as 15 dB. Longer paths may suffer worse antenuation. With this antenuation, in addition to the water on the radomes, the fade margin is cut down to 5 dB.

Glass

For stations that are going to mount the RF/Dish assembly inside, the loss through the glass must be taken into account. A station in Boston is using the MA/COM for a video link. The path distance is 1000 feet. But because the transmit antenna is located inside a building that has silver coated mylar inside the windows, the estimated path length is approximately 10 miles. Asking for an exact figure of glass loss does not give a black and white answer. Clear glass may have a loss of 2 or 3 dB, while mylar coated windows may have a loss in excess of 50 dB. Is the glass double or triple pained? Smoked glass typicially has a loss of 6 to 8 dB. So there is no exact answer.

Delay

Because the digital encoding/decoding process takes time, a small delay is developed. Measured on the WMJX system, (fig. 14), the delay runs approximately 150 micro seconds. Of the air personal, only two people seem to notice the delay to any degree, and even they are adjusting to it. As a side note, we have found that when the station has had to go back to the phone lines, the air staff starts to complain about the "crummy sounding phone lines", and will ask when they can switch back to the STL system. It did not take long for the air staff to become "spoiled" to the high quality of the STL system.

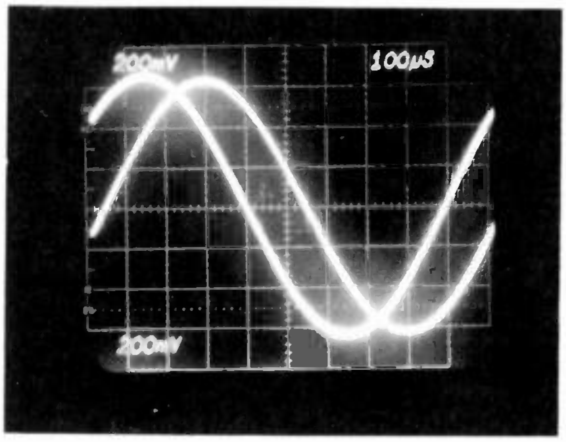

Fig. 14

Backup

Any STL system can fail, and this system is no exception. Especially if a low fade margin is being maintained. This is why stations should maintain a backup. In addition to the two 23 GHz STL systems at WMJX, the station also maintains two sets of stereo telephone lines. In case of an STL failure, the air staff can go to either the second STL, or the phone lines. Going one step farther, an audio silence sensor, in conjuction with an audio switcher, was set up as an automatic audio switch. If the output of the audio switch fall silent, this has to happen for 30 seconds, the silence sensor will trigger a shift register which, in turn, will start the audio switch "looking" for an active audio source. Once an active audio source is located, the silence sensor resets and normal program is restored. This system has worked very well in the past when one STL failed. 30 seconds later, audio was restored, automatically.

CONCLUSION

As 950 MHz become even more congested, 23 GHz is becomming a more viable alternative. However, even 23 GHz is starting to become crowded. Broadcasters are not the only people using this band of frequencies. Those of you interested in going with this system should try to make a decision soon. As time progresses, finding an open 23 GHz frequency may in itself become a challenge.

The modifications that were made to the WMJX system are only a guideline. More than likely, no modifications will have to be made on your system. There are many of these systems in operation throughout the country that are working without being modified.

Finally, there had been some rumors that dbx had discontinued the 700. The dbx 700 is currently out of production, but dbx is actively supporting the unit.

ACKNOWLEDGEMENTS

There are many people to whom I am indebted to for their time and help in preparing this paper. Thanks to Kevin McNamara, Chief Engineer at WMJX/WMEX, Norman Gagnon, Assistant Chief Engineer at WMJX/WMEX and David Wing, Engineering assistant at WMJX/WMEX. Thanks also to Milford (Smitty) Smith, Vice President of Radio Engineering at Greater Media for his assistance.

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dbx, Inc.

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MA/COM, Inc.

Jerry Reese, Chief Engineer, LOCATE

Figure 2 Courtesy of MA/COM, Inc.

Figure 4 Courtesy of dbx, Inc.

REFERENCES

1) Gary M. Miller, Modern Electronic Communication, Prentice Hall, pp. 358-360

2) Ibid., pp 364-365

3) Robert W. Adams, Model 700 Digital Audio Processor Instruction Manual dbx, p. 21

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MICROWAVE ALTERNATIVES FOR RADIO STL

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Abstract
The increasing demand for additional STL frequencies is not new to us. Several alternatives have been suggested in the past few years including the use of 23 GHz Microwave for radio STL. 23 GHz offers a high quality means of transporting FM Stereo, AM Stereo, SCA and other signals over a single microwave link. This paper will discuss the planning, pitfalls, installation and system performance and reliability.

INTRODUCTION

This paper is based on the system in use at KZLA/KLAC, Los Angeles using a M/A-Com MA-23CC 23 GHz microwave with six foot diameter Mark Antenna Products dish antennas at each end of an 8.5 mile path. The M/A-Com microwave has input/outputs for one video channel and up to three audio channels.

The wide bandwidth of the video channel provides the ideal medium for delivery of the FM Stereo programing, which is first digitized and converted to a composite video signal using a DBX Model 700. In this installation the AM Stereo program is transported on two of the microwave's audio channels. One additional audio channel could be used for SCA delivery or other purposes, as needed, but is not installed in the particular system.

WHY 23 GHz.

The demand for 950 MHz STL channels has exceeded the availability. In many markets channels just out of band were used on a "Special Temporary Authorization" (STA) basis. The FCC has made some of these permanent but this only provides help for a few markets. The FCC has also allowed the use of portions of the TV ENG bands on a shared basis. In major markets these frequencies are fully used for Electronic New Gathering.

As microwave technology raised the upper limits of practical system usage, the FCC opened the 23 GHz region for private fixed microwave. Since almost anyone could now have a microwave for almost any good reason, radio broadcasters may also avail themselves of this new piece of spectrum.

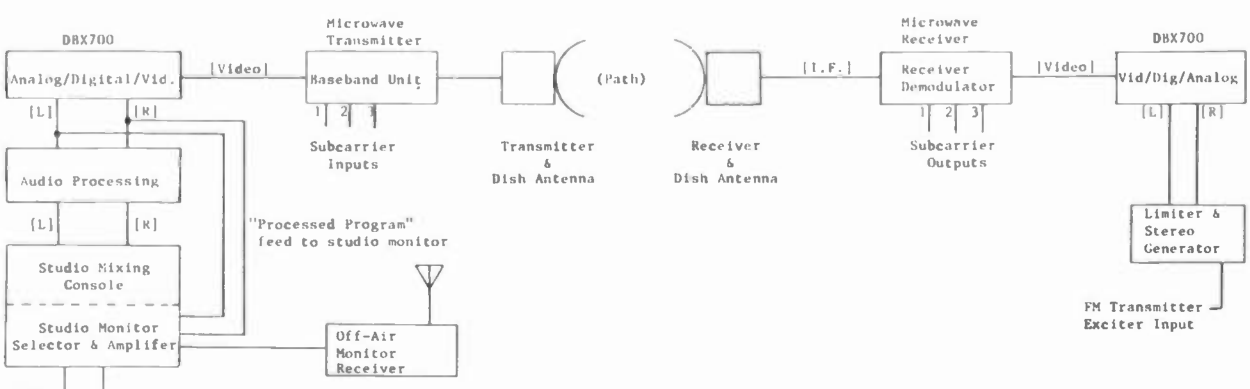

Fig. 1
Digital Microwave Block Diagram

23 GHz Characteristics

Being true microwave, transmission at 23 GHz is very much line of sight. With a line of sight path available, many factors affect the performance. These include antenna gain, distance, free space path loss, precipitation, fog or smog, even the oxygen content of the air. The three determining factors in the workability of the path with which we are most concerned are distance, dish size and the amount of precipitation common to the area.

Where distances approaching 20 miles are common for 950 MHz STL, at 23 GHz the distances for a single hop are limited to about 10 miles for four foot dishes. Using six foot dishes will allow up to 15+ miles under good conditions. In the summer of 1985 one test was conducted in Los Angeles using two foot dishes over an 18 mile path. This test was from Hollywood up to Mt. Wilson. Smog was moderate with a varying inversion layer. This system worked on that day with signal levels several db above the minimum to operate. I suspect that this path could be operated successfully using six foot dishes except during heavy rain.

One of the many factors effecting this frequency band is path loss. Eq. 1 [1] shows the path loss for any frequency and distance.

$$\text{Path Loss} = 36.6 + 20\log F + 20\log D \qquad \text{Eq.1}$$

Where D is the distance in statute miles, Frequency, F, is in MHz.

Using this equation, the path loss for an 8.5 mile 950 MHz path is -114.7 db. If 23 GHz is used over the same path the loss is -142.42 db!

The next greatest contributing factor is precipitation. The rain rate determines the rain drop sizes and thus the attenuation. The greater the rain rate, the larger the drops will be. Thunderstorms will produce heavy rain. The storm cells are generally small. The attenuation will thus not be uniform along the path. A light or steady rain, not from such a violent storm, would produce less attenuation but over the greater path. Experience in Los Angeles has shown that only very heavy rains cause outages over an 8.5 mile path. Light rains will reduce the signal strength some, but not enough to squelch the receiver.

Attenuation due to rain is as follows:

Table 1. [2]

ATTENUATION DUE TO PRECIPITATION
(at 23 GHz)

Precipitation (in./hr.)	Attenuation db/km	Attenuation db/mi
.05	.120	0.075
.10	0.265	0.165
.50	1.621	1.008
1.0	3.462	2.153
2.0	7.60	4.72
4.0	15.0	9.33
5.0	17.8	11.0

To determine the budget for a path we add these losses and receiver noise in db. Subtract them from the sum of the transmitter output in dbm and the antenna gain for both transmit and receive dishes in dbi. Losses due to oxygen could also be considered. This is small, only 0.0124 db/mile.

For example, take a path of 8 miles using a 4 ft. dish. The budget for this path is as follows:

Power output in dbm.	18.0
Tx. Antenna Gain, dbi	45.5
Path Loss from Eq.1	-141.9
Oxygen Absorption	- 0.1
Rain Attenuation (none)	0.0
Rec. Antenna Gain, dbi	45.5
Receiver Noise	- 12.0
Total Path:	- 45.0

From the manufacturer's technical information [3] we obtained the power output, antenna gains and receiver noise. We also may find a figure for Receiver Threshold. The difference between the net path as figured above and the receiver threshold is the fade margin. This determines the amount of signal that can be lost due to rain or any other reduction in signal to the receiver before service is lost.

Table 2
23 GHz Path Fade Margins

	2 ft. Antenna			4 ft. Antenna			6 ft. Antenna		
(Miles)	No Rain	.5"/hr.	2"/hr.	No Rain	.5"/hr.	2"/hr.	No Rain	.5"/hr.	2"/hr.
1.0	36.2	35.1	31.4	47.2	46.1	42.4	54.2	53.1	49.4
2.0	30.1	28.0	20.7	41.1	39.0	31.7	48.1	46.0	38.7
4.0	24.1	19.8	5.2	35.1	30.8	16.2	42.1	37.8	23.2
8.0	18.0	9.5	-19.8	29.0	20.5	-8.8	36.0	27.5	-1.8
16.0	11.9	-5.1	-63.7	22.9	5.9	-52.7	29.9	12.9	-45.7

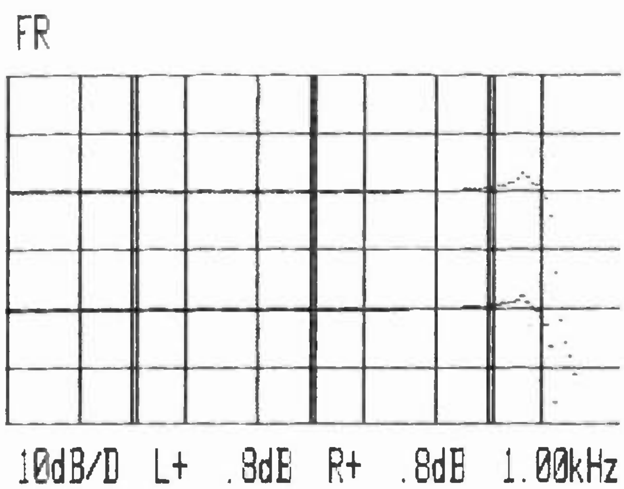

Fig. 2
Frequency Response--DBX700

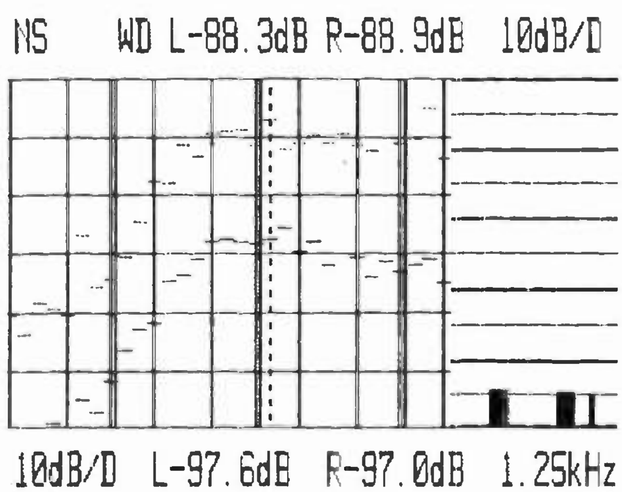

Fig. 3
Noise Response--DBX700

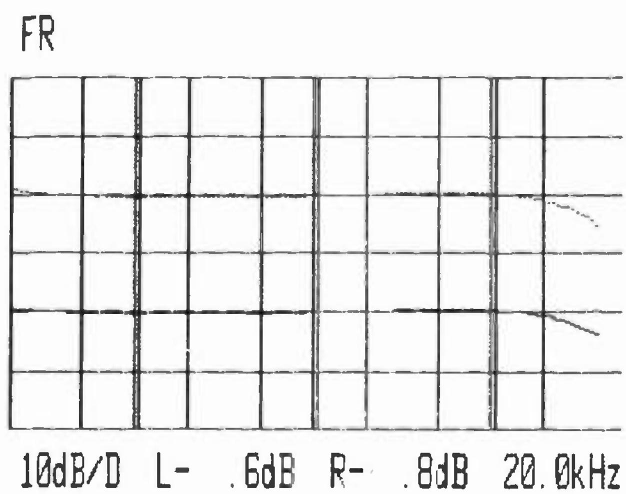

Fig. 4
Subcarrier Frequency Response

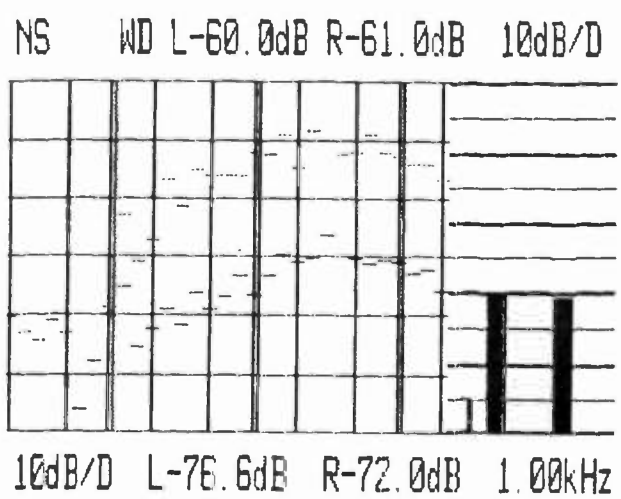

Fig. 5
Subcarrier Noise Response

Table 2 shows the fade margin for 2, 4 and 6 foot dishes for paths 1 to 16 miles. Calculating path loss is a good exercise but is not usually required. Most manufacturers provide charts showing link availability.

USE OF VIDEO & SUBCARRIERS

Channel occupancy of a 23 GHz system (simplex) is 10 MHz. The Video is full color bandwidth. The audio subcarriers used by M/A-Com are at 6.2, 6.8 and 7.5 MHz. A 10 MHz baseband version can carry one video and three audio subcarriers or one video, one audio and one T1 data link.

One test performed at KZLA was to use the video channel as a conventional composite baseband for FM stereo. Performance was adequate to pass an FM proof as formerly required by the FCC. Response was flat and the noise was at -63 db. This is acceptable but nothing to brag about. This is the rationale behind using digital systems with the video channel.

Digital Performance

Several products have been on the market designed to record digitized audio on video tape. Each uses their own format of Pulse Code Modulation. One such unit is the DBX-700. This occupies 5.25 inches of rack space. Two audio modules are located on the left side of the chassis. The left is used for Left and Right stereo channels which are digitized and combined on a single video channel. Next to the encoding module is the playback, or decoding module. For the originally intended application, that is, recording high quality audio on video tape, the same chassis unit is used for recording and then for playback. Separate balanced XLR connectors are used for input and output. Separate BNC connectors serve video out for recording and input for playback.

Using the DBX-700 with the microwave requires two units. One is used to encode the audio into video. This is located at the studio (or source end of the path) and the other at the transmitter. If multiple hop paths are required, the video output of the first receiver simply feeds the video input of the second transmitter, and so on. The decoding DBX-700 is located at the final destination, the stations transmitter site. See Fig. 1.

While we have discussed that the subcarriers could be used for a multitude of purposes, we are mostly interested in the performance of the main "video" channel as used for our main programing channel. What makes this work so well is that the microwave radio system is transparent to the audio performance. The audio quality is that of the digital system.

When used in the decoding mode at the receive site, the digital unit checks the data for errors. An error correction scheme is employed to correct data if necessary. This comes into use when, due to rain, the path starts to approach the receiver threshold, and fades. The error correction will provide useful audio down almost to the noise.

Figures 2 & 3 show the frequency response and noise. These figures were taken using a Sound Technology 1510A test system. Overall noise is rated by the 1510 as -88.9 db! This is the improvement over using 950 MHz STL's or the video channel as a composite STL.

Subcarrier Performance

The signal generated by the DBX-700 is video and does contain sync pulses. The frames however are not continuous analog video. They are made up of digital pulses. These square waves are rich in harmonic content. Thus the bandwidth of the DBX-700 output is much greater than standard picture video. The video baseband of the microwave is wide enough to accommodate up to 10 MHz. (12 MHz available as an option.) This causes the wide bandwidth of the digital "video" to overlap the audio subcarriers residing at 6.2, 6.8 and 7.5 MHz.

Optional video filters are available for both the transmitter and receiver. The installation of one of these filters in the microwave transmitter is essential! It is highly recommended for both ends. Of course this will reduce the higher frequency components of the recovered video. If this were real picture video, this could be a problem. However, remember we are transmitting digital data: square Waves. Even with the attenuated upper end response, the square waves are easily cleaned up in the input circuitry of the digital decoding equipment.

Figures 4 & 5 show the frequency response and noise of two audio subcarriers. It can be seen that they are consistent between channels. Performance of the third, 7.5 MHz, subcarrier would be similar. The overall performance of these subcarriers is as good as some 950 MHz STL's, and better than most phone lines.

Many uses can be found for the subcarrier channels. SCA programing, transmitter remote control, data, comm lines,...and the list goes on. One other use is stereo programing. The performance is much greater than minimum to carry AM stereo audio.

CONTROL & MONITORING

FCC regulations require that each STL be monitored and controlled. The operator on duty must be able to turn the microwave carrier on and off. This is a problem with the MA-23CC. It is lacking a carrier on/off switch. The control cable going from the rack mounted baseband chassis up to the dish carries power for the transmitter which is located at the dish. A simple switch must be added to this B+ line. Unplugging the system from the wall outlet is not an acceptable procedure.

The receiver has a type "c" relay contact brought out to the rear panel. This operates with the squelch control to indicate presence of the carrier. This should be connected to the stations transmitter remote control status inputs so that receiver status can be monitored at the studio. In addition, this relay contact can be used to control a relay which will automatically switch the audio from the microwave to backup phone lines or 950 MHz STL in the event of a path fade due to weather, or simply a failure of the equipment. If this is done, a time delay should be connected such that the audio will not be switched back to the microwave until the system has been stable for at least five minutes. If the outage is due to rain, the squelch may open and close several times per minute. Without the delay, the relay switching the audio could chatter and noticeable clicks and pops will be heard on the air.

The one negative aspect of this STL system is delay. The conversion of analog stereo audio to digital and then back again takes time. This delay is about a third of the delay expected on the round trip to a geostationary satellite. Announcers have great difficulty talking when everything they say is delayed back to their ears. Most announcers also prefer to listen to the off air signal because of the audio processing. Mixing techniques are much different when the audio is compressed.

It is important in most operations for the announcer, who is also the station operator, to monitor the off-air signal to determine quality and simply to know when the station is off. Yet the delay makes headphones hard to use. The solution to this is for the announcer to monitor the off-air signal on the studio speakers as always. A separate feed to the headphones allows the announcer to monitor the audio as it enters the digitizing equipment to feed the transmitter. This way the announcer can hear the air monitor yet is not bothered by the delay in the headphones while on mic.

When the STL is a microwave and not telephone lines, the signal-to-noise performance is great enough to place the bulk of the audio processing at the studio. Since the audio is carried to the transmitter in discrete channels the processing can be placed conveniently for adjustments. The final limiter and stereo generator can remain at the transmitter where composite signal leads can be kept short to maintain good FM stereo separation and quality. This also has the benefit of headphone monitor being as close as possible to the processed sound of the air signal.

FCC LICENSING

When the FCC opened up the 23 GHz band they set licensing up as two classes of service. Private Fixed Operational Microwave under Part 94 of the Rules and Common Carrier Fixed Operational Microwave under Part 21. As a private user, broadcast stations fall under Part 94. The form necessary to apply for a license is FCC Form 402. In item 8 on the 402 form the applicant must specify the "Rule Section Under Which You Are Eligible". There are some 35 catagories of use, each covered by a separate rule. To use 23 GHz for broadcast STL the rule section of eligibility is 90.75(l) -- Business.

Before an application can be filed a frequency must be found. Unlike the 950 MHz STL band, frequency coordination is mandatory. To do the proper coordination on this band the coordinator must have proper and current information on the use of the band in the area. This is beyond the scope of local broadcast frequency coordination since the band usage covers many different industries. This therefore mandates the use of commercial frequency coordinating firms. Many of these operate throughout the U.S.

For Section III of the FCC Form 402 (lines 22 - 72), the frequency coordinator will normally provide either a typed form with these lines properly entered, or a computer printout listing these line. The information from the computer printout could be retyped onto the form: however this is not necessary. Type the word Attached in the blanks at the top of each subsection of Section III with " (quote marks) in the other blanks. Be sure to attach the coordinators printout to the Form 402. The FCC recognizes this as being from the coordinator and correct. A statement from the coordinator must also be attached showing the channel usage in the general area.

INSTALLATION

Before installing the antenna dishes, be sure to check with the local building inspectors office. Permits and inspections may be required. It is common for permits to not be required for mounting a dish on an existing radio tower. However the installation at the studio will most likely require providing a pole or other support structure not presently on the building.

If you plan on the installation of either 4 foot or 6 foot dishes on top of tall buildings have the support designed by a registered mechanical engineer. His plans should be acceptable to the building inspectors. He would also share the liability should the dish come off in the wind due to improper bracing and support, becoming a flying saucer landing several blocks away.

When your frequency is found it will be specified as being either horizontal or vertical. Be sure that the antenna feed horns are mounted for the proper polarization and that the waveguide between the transmitter and feedhorn match. Be sure all connections are tight and waterproof. Include drip-loops in coaxial and control cables. Cables must be secured so as not to move in the wind. These are basic installation practices but are essential at microwave frequencies.

Rack mounted equipment can be mounted at any convenient location. Before mounting equipment in the rack, operate it on the bench. Pay particular attention to the amount of heat given off of each piece. Provide plenty of ventilation when needed.

Digital equipment such as the DBX-700 provides a certain amount of error correction. A LED may be provided on the decoding unit to indicate when this error correction is occurring. If this is operated in locations with high RF levels, RFI may be confused for bad data. Extensive grounding may help. Ferrite beads should be added to the audio leads inside the XLR connectors, and clamp on beads on coaxial cables. The AC power cord should be wrapped several times through a ferrite toroid ring. RFI at your transmitter site could be your worst enemy.

Care must be taken with the audio subcarrier channels. Initial testing was disappointing with poor response and high distortion. This is easily corrected by adjusting the receiver subcarrier tuning to the subcarrier being transmitted. These should also be checked from time to time to keep performance optimum.

When purchasing this type of equipment, be sure to order a technical service manual. Many owners manuals supplied by the manufacture covers installation and use only. Schematics and alignment procedures are not included and must be purchased separately

Once the dish antennas are mounted, they must be adjusted to align them on the direct line of sight path. Each dish may be rotated from left to right and up and down. Connect a DC voltmeter to the AGC line at the receiver. Alternately adjust both the transmit and receive dishes in both the horizontal and vertical planes. At 23 GHz the beamwidth is very narrow. Six foot dishes have half power beamwidths of .5 degrees. Care must be taken to position them correctly. As the dish is moved the voltmeter will peak. Pass the dish through the point of the peak to no signal on either side and then adjust for the maximum. It is very easy to set the position of the antenna to a sidelobe. This will always result in lower signal level. At this frequency we need all the signal we can get. Several adjustments of each dish will be required before the proper setting is reached. Once they are set, tighten the lockdown hardware.

CONCLUSION

While 23 GHz and digital audio is still a relatively expensive technology it offers the broadcast industry a new alternative to traditional studio to transmitter links. Especially in larger markets where the density of 950 MHz is at maximum, 23 GHz should be given serious consideration. The performance and quality can give your station a transparent sound never before heard.

REFERENCES

[1] Eq. 1, Reference Data For Radio Engineers, Sixth edition, 1975 IT&T, Page 28-19.

[2] Reference Data For Radio Engineers, Sixth edition, 1975 IT&T, Table 1 is derived from graph (fig.22), Page 28-18.

[3] RF performance specifications on MA-23CC microwave system are taken from M/A-Com MAC Bulletin 9234C MA-23CC, 1985 M/A-Com MAC, Inc.

FREQUENCY DIVISION MULTIPLEX TO INCREASE THE CAPACITY OF AURAL STL CHANNELS

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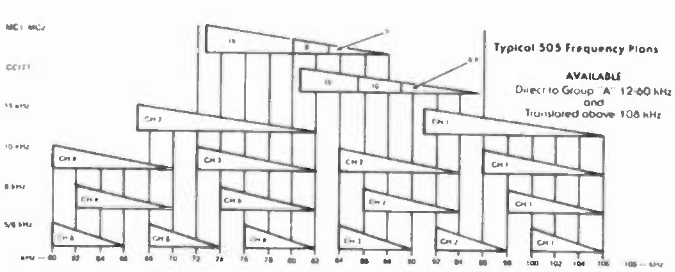
KOA Radio
Denver, Colorado

As congestion becomes a nearly universal problem in the broadcast auxilliary channels methods for maximizing channel loading become a necessity. This paper will discuss the application of Frequency Division Multiplexing to the STL band using commonly available hardware.

INTRODUCTION

Frequency Division Multiplexing has been used for many years by the common carriers to maximize the utilization of wire line and microwave systems. Because the common carriers provided both narrow and wideband audio services the hardware has been developed to stack the channels to achieve any mix of channel bandwidths required over a given path. The basis of Frequency Division Multiplexing is through single sideband techniques that translate the audio to a new frequency.

The most familiar use of this technology in broadcasting is the FM stereo left minus right channel centered on 38 kHz - just above the mono left plus right channel baseband. Since you are already familiar with this use of amplitude modulation as a subcarrier modulated onto an FM system we can step directly up in frequency to the aural STL band.



ADAPTABILITY TO AURAL STL'S

Most composite STL systems are directly plug-in compatible with available Frequency Division Multiplex equipment. Some of the single channel monaural STL equipment can accept one or more frequency division channels above the baseband by use of the existing telemetry or SCA input. Consult the manual or the manufacturer to determine the range of subcarrier frequency that your STL will accept then fill that area with FDM channels. It is possible in most monaural STL units to insert a connector to allow direct input to the modulator in the transmitter and from the detector in the receiver so that one plug will access the whole useable bandwidth.

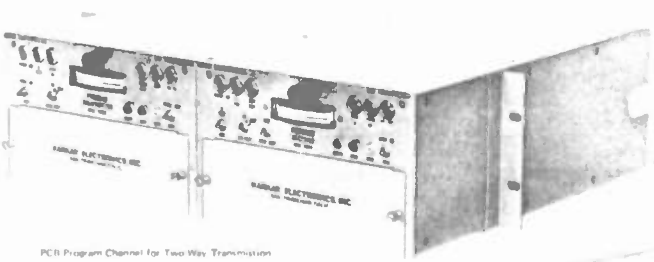
USES OF EXTRA AUDIO CHANNELS

The additional channels opened up by Frequency Division Multiplexing can be used in many ways; The audio for several co-located facilities can be combined to use one STL channel, RPU cueing and dispatch can be relayed to a remote base transmitter, The extra channel can accommodate inter-city relay use freeing up another STL channel or oneleg of a multihop STL may be consolidated with an existing STL freeing up a STL channels. If the same STL transmitter can be received at several points of interest then one STL channel might even be used to deliver program to more than one site along a path. It might even be worth looking for a studio location on the same bearing from both your AM & FM sites!

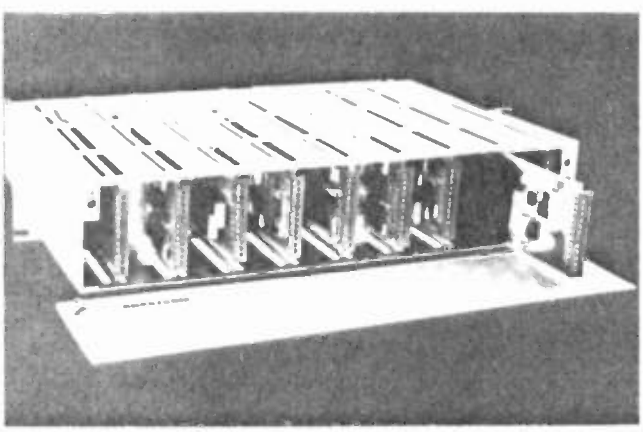
AVAILABLE FDM EQUIPMENT

Two major companies offer FDM equipment off the shelf. Karkar Electronics Inc. of San Francisco, CA offer a complete line of multiplexers for 4 kHz, 5 kHz, 7 kHz, 11 kHz, and 15 kHz audio channels in the PC series. TecTan (previously Coastcom) of Concord, CA offers systems for 5 kHz, 6 kHz, 8 kHz, 10 kHz, and 15 kHz audio channels in the 505 series.

Both companies offer two audio channels per rack shelf with overall costs ranging from three to five thousand dollars per channel depending on options. TecTan offers a special 15 kHz stereo program terminal set Model 522 that is specially designed to maintain stereo phase properly by use of a synchronizing pilot tone.



PCB Program Channel for Two Way Transmission



PERFORMANCE CAPABILITIES

The operating specifications for the FDM units call for frequency response ±0.5dB from 50 kHz to the rated design frequency, harmonic distortion better than 1% at peak program level 10dB above reference, and signal to noise specs well above 50dB and cross talk attenuation well above 50dB. With actual installations in STL service the STL system will probably set the signal to noise and cross-talk limits but a system that met FM stereo proof specs should be very useful in FDM service as well. Additionally DBX or other companding methods should extend the S/N of each channel to exceed broadcast specs by a good margin.

Using the TecTan system as an example a standard composite STL link should be able to convert to provide two 15 kHz program channels plus either two additional 15 kHz channels or four additional 8 kHz channels or six additonal 5 kHz channels or a combination of the above bandwidths.

CONCLUSION

Frequency Division Multiplex can reasonably meet the demand for additional audio channels on our crowded STL band.

SPECIAL REPORTS

Interim Voluntary National Standard
National Radio Systems Committee

Modulation, Overmodulation, and Occupied Bandwidth
Harrison Klein

Amplitude Modulation Theory and Measurements—New and Old Paradoxes
Leonard R. Kahn

Advanced Television Systems
Robert Hopkins

THE NRSC VOLUNTARY NATIONAL STANDARD FOR AM PRE-EMPHASIS

John Marino
NewCity Communications
Bridgeport, Connecticut

William Gilbert
Delco Electronics
Kokomo, Indiana

The past year has been a busy one for the NRSC and especially beneficial to the AM broadcast industry.

A voluntary pre-emphasis/de-emphasis standard was established and now awaits the approval of the American National Standards Institute.

The NRSC actually began its work in September of 1985. By pulling together an active group of broadcasters, receiver manufacturers and audio processor designers, a sub-group was formed to study the pre-emphasis/de-emphasis issue.

We took a systems approach to evaluate what could realistically be accomplished to improve the technical quality of AM.

Our receiver manufacturers detailed the design philosophy that goes into a modern solid state receiver. Reasons for restricted bandwidth were discussed in relation to an AM band crowded with stations and interference.

Looking at some of the circuits incorporated in today's AM auto radios can be awesome to a broadcaster who deals daily with the transmission side of the business. Present receiver technology in the automobile environment is not as simple as many of us once believed.

Audio tapes were recorded showing actual interference conditions which were unacceptable to our listeners. We asked ourselves, "how were these stations ever allowed to be allocated"; driving home the real reason why so many AM stations are in deep trouble today.

Prototype audio processors and receivers were built and evaluated with closed circuit interference simulations. Processors and filters were installed in actual broadcast stations and studied with trained ears.

Finally, a system was developed based upon utilizing a fixed standard pre-emphasis/ de-emphasis curve. This curve, agreed upon by the entire committee, was found to be the best compromise between most of today's narrow bandwidth radios and the new wide bandwidth units.

As NRSC compatible receivers come out of production, broadcasters who follow the guidelines laid out in this standard will discover a vastly improved AM spectrum.

Standardized transmission and reception characteristics are the key to improving the fidelity of AM.

With the filters used in today's radios, there is no fidelity to be gained by excessively boosting high frequencies during the transmission process. We must have complementary receivers.

During the NRSC comment period, which ended in January of this year, nearly all responses to the standard were positive. The few negative comments dealt mainly with our proposed 10 kHz bandwidth recommendation.

Please remember that we, as broadcasters, allowed the AM service to deteriorate. While we became pre-occupied with FM, new AM technology stagnated.

AM radio can be improved by taking a standard systems approach. Receiver manufacturers will produce better radios, but only if we do our part. Ten kilohertz audio will gain us a lot more listeners than three kilohertz audio.

Additionally, ten kilohertz audio, through the NRSC compatible system, will produce a lot less interference than fifteen kilohertz audio that cannot be received on most of our present day radios.

As you read, and ultimately implement the standard, remember that you are taking one more step toward rebuilding AM.

Never has there been so much support for this effort, not only by joint industry groups, but also by the National Telecommunications and Information Administration, which is urging all broadcasters to operate in accord with the standard.

This is our opportunity, in an age of deregulation, to voluntarily join forces and work toward AM improvement.

NATIONAL RADIO SYSTEMS COMMITTEE

Electronic
Industries
Association

National
Association
of Broadcasters

INTERIM
VOLUNTARY NATIONAL STANDARD

1. 75 µS AM Broadcast Transmission Preemphasis
2. Complimentary 75 µS AM Receiver Deemphasis
3. 10 kHz AM Transmission Bandwidth
4. Five-year Review Provision

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Kokomo, Indiana

January 10, 1987

Sponsored by the Electronic Industries Association and the National Association of Broadcasters

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§ 1. SCOPE

The National Radio Systems Committee (NRSC) is a joint Committee composed of all interested parties including representatives of AM broadcast stations, AM receiver manufacturers, and broadcast equipment manufacturers. This document describes an interim[1] voluntary national standard that specifies the preemphasis of AM broadcasts, the deemphasis of AM receivers, and the audio bandwidth of AM stations prior to modulation. The standard applies to AM monophonic and AM stereo L+R transmissions, and to dual bandwidth and single bandwidth AM receivers. Compliance with the standard is strictly voluntary. To the NRSC's knowledge, no industry group or entity is or will be adversely affected by issuance of this document. Every effort has been made to inform and accommodate any and all interested parties. The NRSC believes that implementation of the standard will reduce AM interference, increase useful AM service areas, and encourage the production of higher fidelity AM receivers.

A five year review provision is established.

§ 2. INTRODUCTION

On September 5, 1985, the NRSC adopted a resolution to study proposals to standardize AM transmission preemphasis and AM receiver deemphasis with the objective of establishing an industry-wide AM preemphasis/deemphasis voluntary standard. After twelve months of study, on September 10, 1986 the NRSC released a draft voluntary standard that proposed a specific AM preemphasis/deemphasis curve as well as a 10 kHz standard AM bandwidth. The bandwidth specification evolved from NRSC deliberation on the causes and cures of AM interference, and ways to technically encourage the production of higher fidelity AM receivers. After a three month public comment period, the NRSC, on January 10, 1987, formally adopted this standard and authorized its publication by the National Association of Broadcasters and the Electronics Industries Association.

The purpose of the NRSC voluntary standard is to create a transmission/reception system where (1) AM broadcast stations will know, with certainty, the likely audio response characteristics of AM receivers, and (2) AM receiver manufacturers will know, with certainty, the likely audio response characteristics of AM broadcasts. A "matching" of preemphasis and deemphasis is expected to improve the consumer's overall satisfaction with the technical quality of listening to AM radio. The NRSC believes that the public interest is served by establishing a compatible transmission/reception system and the accompanying improvement of AM broadcasts and AM receivers.

This document also describes a specification for the maximum audio bandwidth transmitted by AM broadcast stations. Implementation of a bandwidth specification will reduce second-adjacent channel interference and thereby lead to (1) a significant reduction of second-adjacent channel interference as perceived on "wideband" AM receivers; (2) a corresponding increase in the interference-free service areas of AM stations; and (3) an incentive for the further building of dual bandwidth AM "wideband" receivers.[2] Analysis by a subgroup of the NRSC has shown that there would be little if any detrimental effect on today's "narrowband" AM receivers upon the implementation of this voluntary standard.

§ 3. BASIC DEFINITIONS

A. Preemphasis. The boosting of high audio frequencies prior to modulation and transmission.

B. Deemphasis. The attenuation of high audio frequencies during the process of reception and demodulation.

C. "Narrow" receivers. A subjective term to describe receivers with typical combined RF, IF and AF response characteristics of -10 dB at 4.2 kHz, -20 dB at 6.0 kHz. Response characteristics of narrow AM receivers are known to vary widely.

D. "Wideband" receivers. A subjective term to describe receivers with typical combined RF, IF, and AF response of -6 dB

1. The standard is described as "interim" until it is submitted and approved by the American National Standards Institute.

2. First Adjacent channel interference considerations may continue to discourage the building of single bandwidth "wideband" receivers; however, the extent and nature of this form of interference has not been fully studied by the NRSC.

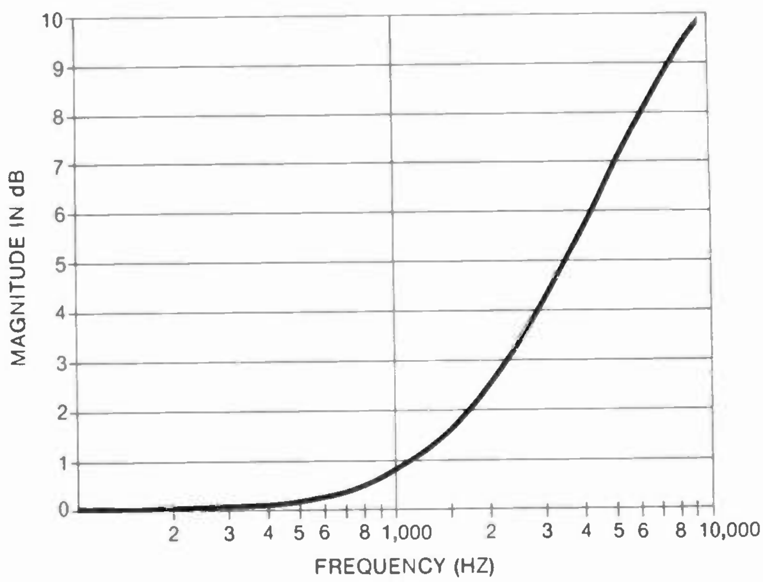

Figure 1. Modified 75μs AM Standard Pre-Emphasis Curve

Technical Information

Frequency	Magnitude (dB)	Phase (deg)	Group Delay (sec)
50	0.00	1.0	-5.6669E-005
100	0.01	2.0	-5.6547E-005
400	0.14	8.0	-5.4175E-005
700	0.42	13.7	-4.9467E-005
1000	0.81	18.7	-4.3318E-005
1500	1.63	25.5	-3.2247E-005
2000	2.54	30.4	-2.2343E-005
2500	3.44	33.6	-1.4509E-005
3000	4.28	35.7	-8.6612E-006
3500	5.05	36.9	-4.4133E-006
4000	5.75	37.4	-1.3702E-006
4500	6.37	37.4	7.8900E-007

Frequency	Magnitude (dB)	Phase (deg	Group Delay (sec)
5000	6.92	37.1	2.3048E-006
5500	7.41	36.6	3.3525E-006
6000	7.85	35.9	4.0592E-006
6500	8.24	35.2	4.5169E-006
7000	8.58	34.3	4.7926E-006
7500	8.89	33.4	4.9357E-006
8000	9.16	32.5	4.9823E-006
8500	9.41	31.6	4.9595E-006
9000	9.62	30.8	4.8871E-006
9500	9.82	29.9	4.7801E-006
10000	10.00	29.0	4.6495E-006

at 6 kHz, -10 dB at 8 kHz. Response characteristics of wide AM receivers are known to vary widely.

E. "Excessive" Preemphasis. Preemphasis that produces no discernable benefit when received by a "narrow" receiver but increases interference to adjacent channel AM stations.

§ 4. AM TRANSMISSION PREEMPHASIS

§ 4.1. In General. AM preemphasis is the boosting of high audio frequencies prior to modulation and transmission. Today, most AM stations use preemphasis to varying extents. This preemphasis is employed in an attempt to compensate for the "narrow" response of most AM receivers. If AM preemphasis is not controlled, one station may interfere with AM receivers listening to neighboring stations located on adjacent AM channels. Whether such interference is objectionable will depend on (1) the response characteristics of the AM receiver, (2) the amount and nature of transmission preemphasis, (3) the extent to which the AM station is employing compression/ limiting techniques, and (4) whether the AM transmission system is bandlimited in the audio processor, transmitter or antenna.

Preemphasis is useful for improvement of the AM transmission-reception system audio response only to a limited extent for receivers using IF transformers. Many receivers using ceramic filters with narrow response characteristics can not be improved by use of excessive preemphasis. These receivers can not "hear" the transmission of preemphasized high audio frequencies. But excessive preemphasis will foster adjacent channel interference and cause wideband radios to sound shrill or strident.

§ 4.2. Description of the Modified 75 uS Preemphasis Curve. Each AM broadcast station shall broadcast with audio preemphasis as close as possible (within the capabilities of the station's transmission system) to the recommended standard, without exceeding it. The curve applies for audio frequencies up to 10 kHz.

The NRSC proposed standard AM transmission preemphasis curve is shown in Figure 1. The curve describes the recommended net transmission system static audio response of an AM station.

The recommended preemphasis curve is a single zero curve with a break frequency at 2122 Hz. It is similar to the 75 uS curve used for FM broadcasting. To reduce the peak boost at high frequencies, a single pole with a break frequency of 8700 Hz is employed. NRSC analysis has shown that the proposed curve is compatible with most existing AM receivers.

§ 4.3. Methods for Determining Performance. The NRSC AM preemphasis curve is a static curve, and cannot be measured dynamically. NRSC studies have shown that the dynamic and non-linear functions performed by most AM station audio processors will modify any given preemphasis curve. In addition, it is the audio response of the entire AM transmission system that indicates performance in accordance with the standard. For these reasons, measuring a station's preemphasis curve for the purpose of determining compliance with the NRSC standard shall be performed in accord with the following specifications:

§ 4.3.1. Use of Audio Tones. Compliance with the curve shall be measured by sweeping the station's transmission system with audio tones. The dynamic functions of the AM station's processor, but not the frequency shaping circuits, must be disabled (i.e., in "proof" mode).

§ 4.3.2. Location of Measurement. The net transmission system audio response is best measured by detecting the over-the-air signal. This will ensure that the AM transmitter and antenna combination is faithfully reproducing the preemphasized audio.[3] Alternatively, if the transmitter and antenna combination is reasonably broadband, performance can be determined by static measurement of the audio signal prior to modulation.

§ 5. AM RECEIVER DEEMPHASIS

§ 5.1. In General. Receiver deemphasis results from the selectivity

3.However, the deemphasis characteristics of the device used to demodulate the AM transmission must be accounted for. Additionally, some AM stations with transmitter or antenna problems may not be able to pass preemphasized audio without introducing "splatter" interference and/or overmodulation. If a particular AM station transmission system cannot "handle" the NRSC recommended curve, it is suggested that a lower amount of preemphasis be used until the system problems are corrected to allow the NRSC curve to be faithfully implemented.

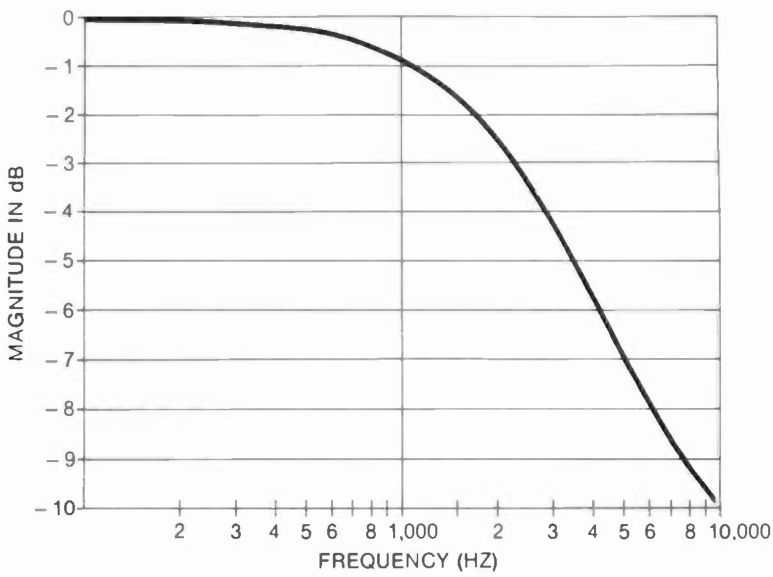

Figure 2. Modified 75µs AM Standard De-Emphasis Curve

characteristics of a receiver's RF and IF stages and the response characteristics of the receiver AF section. A standard deemphasis curve permits AM stations to know, with certainty, the likely overall response characteristics of AM receivers.

§ 5.2. Description of the Standard Deemphasis Curve. AM receivers shall complement the recommended transmission preemphasis characteristic described in § 4 by incorporating a net receiver system audio response described in Figure 2. (The net system audio response of an AM receiver is the combined RF, IF, and AF audio response.) The NRSC deemphasis curve is characterized by a single pole at 2122 Hz and a single zero at 8700 Hz. It is the precise complement of the preemphasis standard described in § 4. The preemphasis/deemphasis voluntary standards apply only for audio frequencies below 10 kHz; the implementation of preemphasis/deemphasis standards produces a transmission/reception system that is essentially flat to nearly 10 kHz and limited only by the AM receiver's choice of bandwidth.

§ 5.3. Methods for Determining Performance. The deemphasis characteristic shall be determined by measuring the overall frequency response in accordance with International Electrotechnical Commission ("IEC") Publication 315.3, Clause 11.2:

(1) The receiver is brought under standard measuring conditions and the reference audio-frequency output voltage is noted. The modulation frequency is then varied and the output voltage at each frequency is noted and expressed in decibels relative to the reference voltage.

The modulation depth is adjusted at each frequency in accordance with the preemphasis characteristic of AM broadcast transmission. To avoid overmodulation at some frequencies it may be necessary to use a modulation factor of less than 30% at some frequencies.

(2) If overloading of the AF section of the receiver occurs, either the volume control attenuation should be increased or the modulation factor reduced, and a corresponding factor applied to the results.

(3) The measurements may be repeated with other values of RF input signal level and frequency.

The frequency response shall be measured for both monophonic and stereophonic reception, in accordance with the definition of the particular AM stereo system. For dual bandwidth receivers, the frequency response shall be measured in both bandwidth positions.

Results may be presented graphically, with modulation frequency plotted logarithmically as abscissa and the output in decibels as ordinate.

The frequency response can be stated as follows:

Selectivity	Frequency Response
Narrowband	50 Hz to 5000 Hz +/- X dB
Wideband or Stereo	50 Hz to 10,000 Hz +/- X dB

(Where X is the maximum deviation from the recommended frequency response, and 5000 Hz and 10,000 Hz are example frequency specifications.) The deviation X shall be of as low a value as practical. If a notch filter is used while the AM receiver is under test, the stated frequency response should be modified accordingly. Suggested modifications include (1) adding an appropriate footnote to the frequency response specification; and/or (2) lowering the upper limit value to the above "wideband" audio response specification.

§ 5.4. Notch Filters. A notch filter is a very selective filter that attenuates the spectrally pure carriers of first adjacent channel AM stations. Although an optional enhancement to an AM receiver, using notch filters is recommended. If used, the notch filter should (1) have as high a "Q" as is practical, (2) adequately suppress the interfering carriers, and (3) not unduly degrade the desired bandwidth performance of the AM receiver.

§ 6. 10 KHZ BANDWIDTH FOR AM TRANSMISSION

§ 6.1 In General. Each AM broadcast station shall modulate its transmitter with an audio bandwidth described by the specification in Figure 3. Appropriate and carefully designed audio low-pass filters as the final filtering prior to modulation can be used to implement this specification. The purpose of the bandwidth specification is to remove interference by controlling the occupied RF bandwidth of AM stations.[4]

§ 6.2. Description of the Standard. The audio bandwidth transmission standard is specified in Figure 3. The audio

4. It should be noted that the operation of non-linear AM Stereo systems theoretically may produce phase modulation components outside the desired RF bandwidth. The NRSC will examine this phenomena with the goal of determining whether such components exist and, if they do exist, whether they are objectionable. For a discussion of the detrimental effect of high audio frequencies on occupied RF bandwidth, see Klein, Modulation, Overmodulation, and Occupied Bandwidth: Recommendations for the AM Broadcast Industry, Proceedings of the 1987 NAB Engineering Conference, Dallas, Texas (April, 1987).

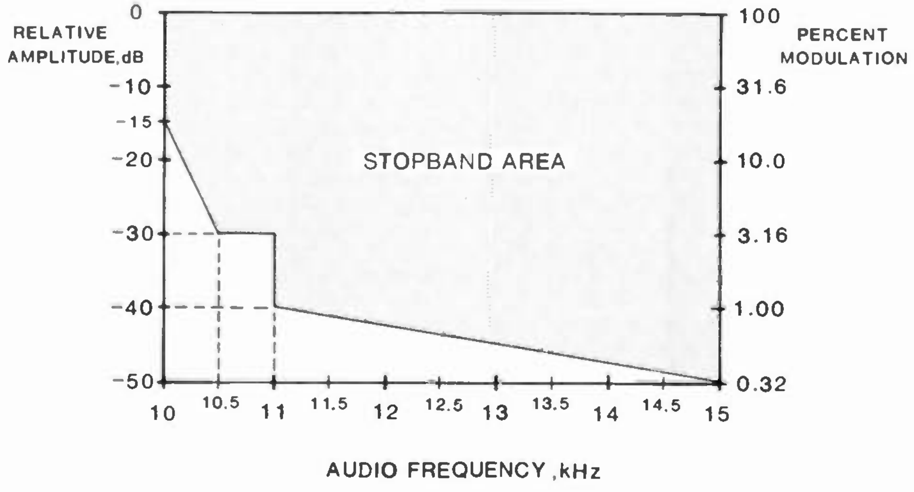

Figure 3. NRSC Stopband Specification
(Audio Envelope Input Spectrum To AM Transmitter)

envelope input spectrum to the AM transmitter shall be -15 dB at 10 kHz, smoothly decreasing to -30 dB at 10.5 kHz, then remaining at -30 dB from 10.5 kHz until 11.0 kHz. At 11.0 kHz, the audio bandwidth shall be -40 dB, smoothly decreasing to -50 dB at 15 kHz. Above 15 kHz, the audio bandwidth shall remain at least -50 dB. The reference level is 1 dB above a 200 Hz sine wave at 90% negative modulation. See Figure 3.

§ 6.3. Method for Determining Performance. An AM station is determined to be in compliance with the NRSC bandwidth characteristic by measurement of the station's audio bandwidth in accordance with the following parameters:

§ 6.3.1. Location of Measurement. Audio bandwidth measurements shall be obtained at the audio input terminals to the AM transmitter. For AM Stereo stations, audio bandwidth shall be measured at the L+R audio input terminals to the RF modulator. Note that the NRSC bandwidth standard characterizes an audio bandwidth that represents station program material that has been modified by possibly non-linear circuits in the station's audio processor. For this reason, the NRSC recommends use of a test signal that adequately characterizes typical audio program material, rather than relying on static audio test tones. However, it may still be useful to measure bandwidth statically at the time that AM preemphasis is measured.

§ 6.3.2. Use of Standard Test Signal. Audio bandwidth shall be measured using a test signal consisting of USASI (United States of America Standards Institute) noise that is pulsed by a frequency of 2.5 Hz at a duty cycle of 12.5%. See Figure 4. USASI noise is intended to simulate the long-term average spectra of typical audio program material. Pulsing of the noise is intended to simulate audio transients found in audio program material. USASI noise is a white noise source[5] (i.e. noise with equal energy at all frequencies) that is filtered by (1) a 100 Hz, 6 dB per octave high-pass network and (2) a 320 Hz, 6 dB per octave low-pass network. See Figure 4. A pulsed USASI noise generator is shown in Figures 5 and 6. Using the attenuator pad, the ratio of peak-to-average amplitude shall be 20 dB at the audio output of the pulser. The station's audio processor must be in normal operating mode.

§ 6.3.3. Use of Standard Measurement Devices. A suitable swept-frequency or FFT (Fast Fourier Transform) spectrum analyzer shall be used to measure compliance with the NRSC bandwidth specification.

(a) Spectrum Analyzer Setup. When a swept-frequency audio spectrum analyzer is used to measure compliance with the NRSC bandwidth specification, the analyzer's setup shall consist of:

a. 300 Hz resolution bandwidth.[6]
b. 2 kHz/horizontal division.
c. 10 dB/vertical division.
d. Reference: 1 dB above 200 Hz (sine wave) 90% negative modulation.
e. Display: maximum peak hold (or equivalent function).

The analyzer's operating span and sensitivity are adjusted as necessary to determine compliance.

(b) Fast Fourier Transform Analyzer. When a FFT analyzer is used to measure compliance with the NRSC bandwidth specification, the analyzer's setup shall consist of:

a. Reference: 1 dB above 200 Hz (sine wave) 90% negative modulation.
b. Window: Hanning.
c. Horiz. span: 20 kHz.
d. Dynamic range: 80 dB or available range.
e. Display: Maximum peak hold (or equivalent function).

5.Acceptable white noise sources include GenRad Models 1382 and 1390B; Bruel & Kjaer Model 1405; and National Semiconductor IC No. MM 5837N.

6.Note: if the audio bandwidth under test fails to meet the NRSC specification when a 300 Hz resolution bandwidth is employed, a narrower resolution bandwidth, such as 100 Hz, may be used to determine compliance; however, the sweep rate and the video bandwidth of the analyzer must be adjusted according to the manufacturer's instructions in order to assure accurate representation of the resolution bandwidth employed. If in doubt, check with the analyzer operating manual or the manufacturer. Further Note: the NRSC may suggest a different or more precise measurement standard as the industry gains experience. Spectrum analyzers that are capable of 300 Hz resolution bandwidths at audio frequencies include, but are not limited to, Tektronix Models 5L4N and 7L5; Hewlett-Packard Models 3580A, 3582A, 3585A, 8553A or B, 8566A or B, 8568A or B, 71100A; Marconi Models 2370, 2382; Rhode/Schwarz/Polarad Model CSA-240M; and a Techron Model TEF System 12.

§ 7. FIVE YEAR REVIEW PROVISION

It is the goal of the NRSC to increase the fidelity of the AM transmission and reception system from its present state to a quality level that approaches the quality available via FM broadcasting. Towards this end, the voluntary standards described in this document shall be in effect for five (5) years from the effective date of this standard. During the interim five year period, this voluntary standard will be reviewed at least once a year to determine whether the fidelity goals of the NRSC are being realized.

§ 8. EFFECTIVE DATES

These dates serve only as objectives. AM Broadcast stations and AM receiver manufacturers are expected to make a good faith effort to implement the NRSC standard.

A. AM Broadcast Stations. The effective date of this standard is January 10, 1987.

B. AM Receiver Manufacturers. The effective date of this standard is January 10, 1988.

§§§§

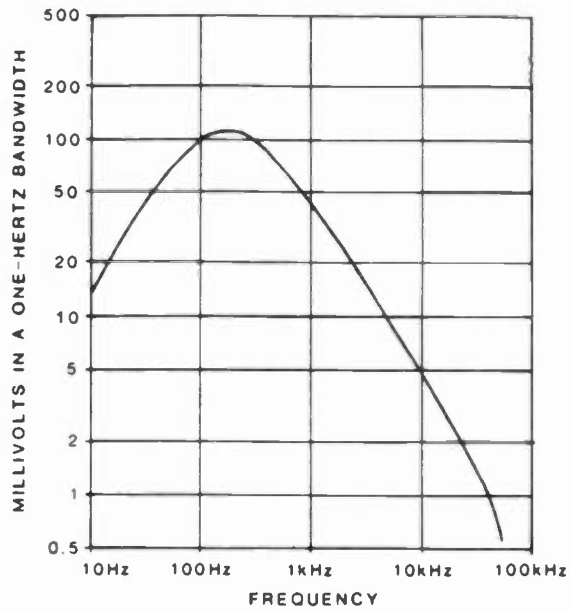

Figure 4. Spectra of USASI Noise

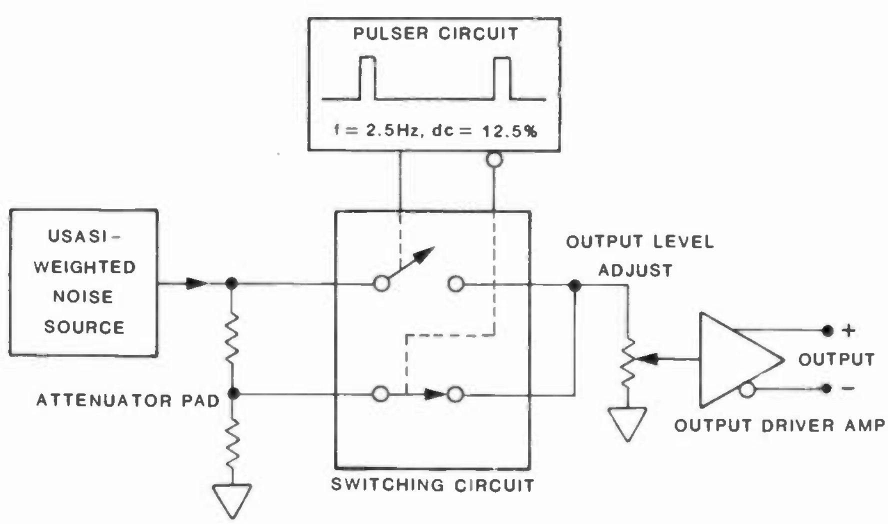

Figure 5. Pulsed-USASI Noise Generator

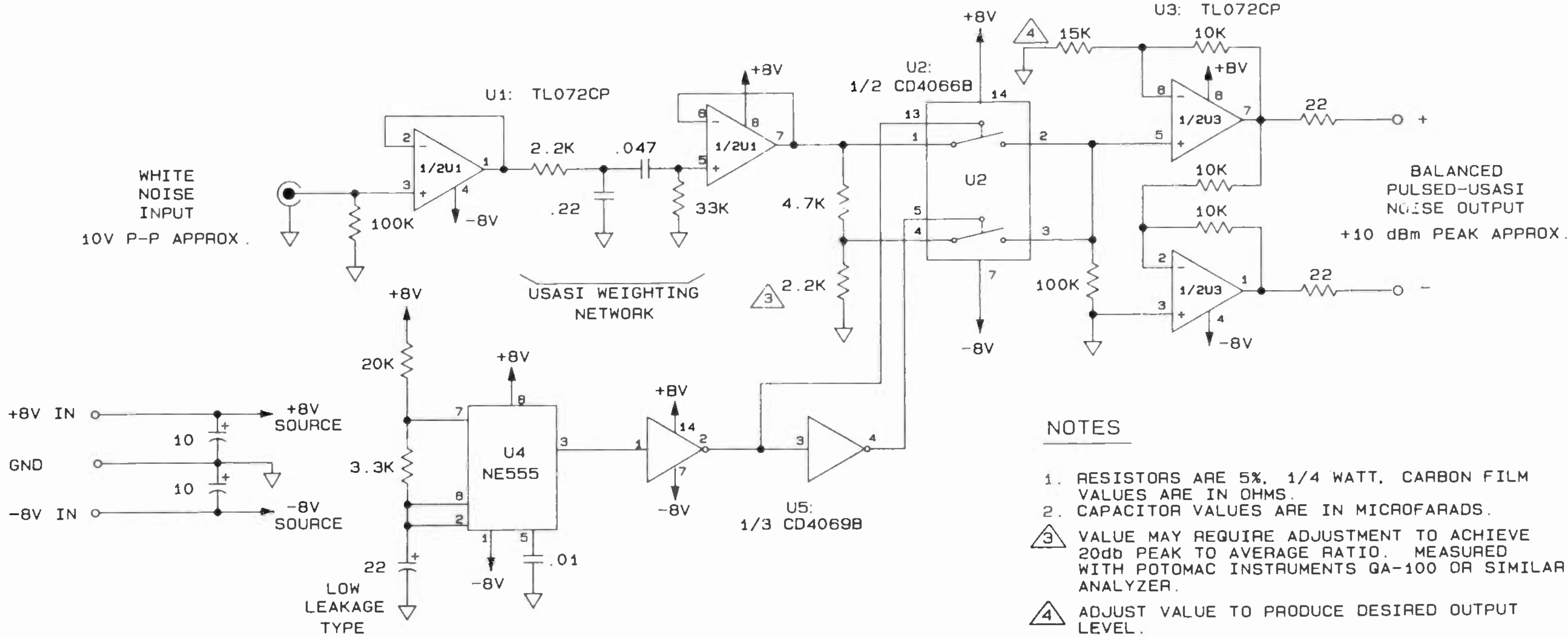

Figure 6. Application Circuit: USASI Noise Weighting/Pulser Circuit

Modulation, Overmodulation, and Occupied Bandwidth:

Recommendations for the AM Broadcast Industry

An AM Improvement Report
from the
National Association of Broadcasters

September 11, 1986

Harrison J. Klein, P.E.
Hammett & Edison, Inc.
Consulting Engineers
San Francisco

on behalf of the
AM Improvement Committee
Michael C. Rau, Staff Liaison
National Association of Broadcasters
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Modulation, Overmodulation, and Occupied Bandwidth:
Recommendations for the AM Broadcast Industry

HARRISON J. KLEIN, P.E.

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FIGURES

Modulation, Overmodulation, and Occupied Bandwidth:
Recommendations for the AM Broadcast Industry

HARRISON J. KLEIN, P.E.[1]

I. Executive Summary

This technical report was prepared on behalf of the AM Improvement Committee of the National Association of Broadcasters. Its purpose is to provide recommendations for AM engineers on how to assure (1) the transmission of a clean and full-fidelity AM signal, (2) the prevention of AM overmodulation, and (3) the prevention of "splatter" interference.

A computer analysis of AM modulation and overmodulation was performed to assess the extent that out-of-band emissions result from overmodulation, improperly processed audio, and RF networks in transmitters and antennas. In addition, tests were performed to obtain data at an operating AM station. The effects of antenna bandwidth are discussed, and suggestions are given on the optimal locations for monitoring AM modulation and the best ways to measure occupied bandwidth.

The following six conclusions and recommendations are provided:

1. The primary cause of splatter interference is not the disappearance of the carrier during overmodulation, but instead is the presence of excessive high-frequency content in the audio signal that modulates the transmitter.
2. Meeting FCC bandwidth limits is no guarantee of a "clean" transmitted signal.
3. Splatter interference is minimized by (1) low-pass filters on audio prior to modulation, (2) final protective clippers in processors or at transmitter inputs, and (3) elimination of DC level shift in AM transmitters.
4. Modulation percentage as observed in the field is often inaccurate, differing from the modulation percentage observed at the transmitter, because of the effect of RF networks in the transmitter and the antenna system; to prevent overmodulation a station must have an accurate modulation monitoring point.
5. AM stations should evaluate modulation performance using appropriate analysis techniques, and tailor transmitter modulation adjustments accordingly.
6. The AM industry should develop a high-quality synchronous detector AM demodulator for accurate analysis of modulation characteristics in the field.

Each of these recommendations is discussed herein. Industry-wide attention to these recommendations would raise the quality of AM listening and reduce objectionable interference.

II. Introduction

Interference between stations is a serious problem for AM radio. Interference has been exacerbated by the increasing number of stations and the rising ambient radio-frequency noise level, but a major factor has been the increase in the occupied bandwidth of typical AM stations. Greater occupied bandwidth results in greater adjacent-channel interference, especially as perceived on wider bandwidth receivers. Excessive occupied bandwidth does *not* improve AM reception quality.

A station's occupied bandwidth depends upon its programming, its audio processing, its modulation level, and its transmitter and antenna bandwidths. Occupied bandwidth must be optimized to most efficiently transmit the desired signal to the receiver while minimizing undesired signal components that can cause interference. To optimize its bandwidth, a station must be able to control and monitor its modulation effectively, and must understand how modulation practices affect occupied bandwidth.

For example, overmodulation is one of the basic mechanisms by which an amplitude modulated carrier can be degraded. On the surface, overmodulation appears a straightforward phenomenon which should be easily understood and prevented. Yet many stations regularly overmodulate, some intentionally, implying that the overmodulation mechanisms are not fully understood or that the stations do not believe there are benefits to preventing overmodulation. On the other hand, receiver manufacturers consider overmodulation to be a significant AM problem. In numerous industry meetings, they have indicated their belief that *splatter caused by overmodulation is a major factor preventing the manufacture of improved, high-fidelity AM radios.*

A review of AM modulation, overmodulation, and occupied bandwidth is timely. This report provides an overview of the subject. Modulation and overmodulation are defined and their effects on occupied bandwidth are described. The spectral energies created by "clean" and by overmodulated AM signals are compared. The distribution

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of this energy is discussed. The effects on occupied bandwidth due to audio processors, transmitters, and antenna systems are reviewed, with emphasis on the roles of audio and carrier clipping.

Most importantly, techniques are discussed that engineers can use to prevent excessive occupied bandwidth. Processing techniques for minimizing overmodulation and splatter are described. Guidelines for modulation monitoring are provided. The benefits of clean signals to both broadcasters and receiver manufacturers are demonstrated.

III. Definitions of Modulation and Overmodulation

An amplitude-modulated carrier with a single modulating frequency can be described by the equation

$$e(t) = A(1 + m \cos 2\pi f_m t) \cos 2\pi f_c t, \qquad (1)$$

where A = amplitude of unmodulated carrier,
m = AM modulation index,
f_m = modulating frequency in Hertz,
and f_c = carrier frequency in Hertz.

The AM modulation index is merely the relative level of modulation; the term has a different and simpler meaning for amplitude modulation than it does for frequency modulation. Figure 1 shows four different waveforms corresponding to the above equation with varying values of m: 0.25, 0.5, 1.0, and 1.25.

Although modulation index is related to percentage of modulation (as discussed below), Section 73.14 of the FCC Rules officially defines modulation percentage in terms of the modulated RF envelope:

Positive percentage modulation = 100 (MAX - C) ÷ C,
Negative percentage modulation = 100 (C - MIN) ÷ C,

where MAX = instantaneous maximum envelope level,
MIN = instantaneous minimum envelope level,
and C = unmodulated carrier level (identical to A in equation for $e(t)$ above).

It can be seen from Figure 1 that, up to 100%, modulation index is equivalent to modulation percentage.[2] However, the two terms are not identical. There can be no negative modulation percentage over 100%, according to the above definition, because the envelope can never be smaller than 0. Yet there is no upper limit on the modulation index. As shown in Figure 1d, the carrier does not disappear when $m > 1$ in the equation, but merely "folds over" during negative modulation peaks.

There are two types of overmodulation discussed in this report. AM broadcast engineers are familiar with the type of overmodulation found in a typical AM transmitter and shown in Figure 2: as the audio input level to the transmitter is increased above that necessary to achieve 100% negative modulation, the carrier clips, or disappears, during negative peaks. It is essential to recognize the fundamental difference between this conventional overmodulation, and the overmodulation shown in Figure 1d in which $m > 1$ but no carrier clipping occurs. The second type of overmodulation is produced when an integrated-circuit multiplier is used to generate amplitude modulation. This is an alternative modulation technique used by some commercially available low-power AM transmitters. Amplitude and phase errors in RF networks, examined in Section VIII of this report, can also produce this effect. For clarity, this report will sometimes differentiate between "transmitter-type overmodulation" (Figure 2) and "multiplier-type overmodulation" (Figure 1d). Where "overmodulation" is used without qualification, transmitter-type overmodulation is implied.

Related to overmodulation is the term "splatter." Splatter is loosely defined to be any radio-frequency spectral components that increase the normally occupied bandwidth of the AM signal. Some splatter may be unavoidable or even intentional, such as the increased sideband energy produced by audio preemphasis. Other splatter is unintentional and can be prevented.

IV. The Amplitude Modulation Spectrum

Using trigonometric identities, the amplitude modulation equation (1) can be written to distinguish the frequency components of the signal. For single-tone modulation (using angular frequency ω instead of f in Hertz, where $\omega = 2\pi f$, to simplify the notation):

$$\begin{aligned} e(t) &= A\,(1 + m \cos \omega_m t) \cos \omega_c t \\ &= A \cos \omega_c t + A\, m \cos \omega_c t \cos \omega_m t \\ &= A \cos \omega_c t \\ &\quad + 1/2\, A\, m\, [\cos (\omega_c - \omega_m) t + \cos (\omega_c + \omega_m) t]. \end{aligned}$$

This equation shows the well-known three-component spectrum of an AM wave. For m = 1 (100% modulation) the spectrum contains the carrier component at frequency ω_c and two sidebands, each 6 dB below the carrier, separated from the carrier by the modulating frequency ω_m.

From this equation, it can be seen that an AM spectrum has several notable characteristics. The carrier component does not vary with modulation and the two sidebands are symmetrical about the carrier. Most importantly, the spectrum of the sidebands is the frequency-shifted spectrum of the modulating audio (inverted for the lower sideband). Audio-frequency components present at the input of the transmitter will appear on both sides of the carrier at their same relative levels. This means that under normal modulating conditions (i.e., no overmodulation or other non-linear effects) the sideband energy is determined directly by the modulating audio spectrum. An audio component of 20 kHz will appear 20 kHz on each side of the carrier. If its audio level is 10 dB below 100% modulation, the sideband levels will be 10 dB below 100% modulation (16 dB below carrier). In a modern transmitter with good high frequency response, any high-frequency distortion

[2] This statement is valid for any symmetrical modulating audio waveform.

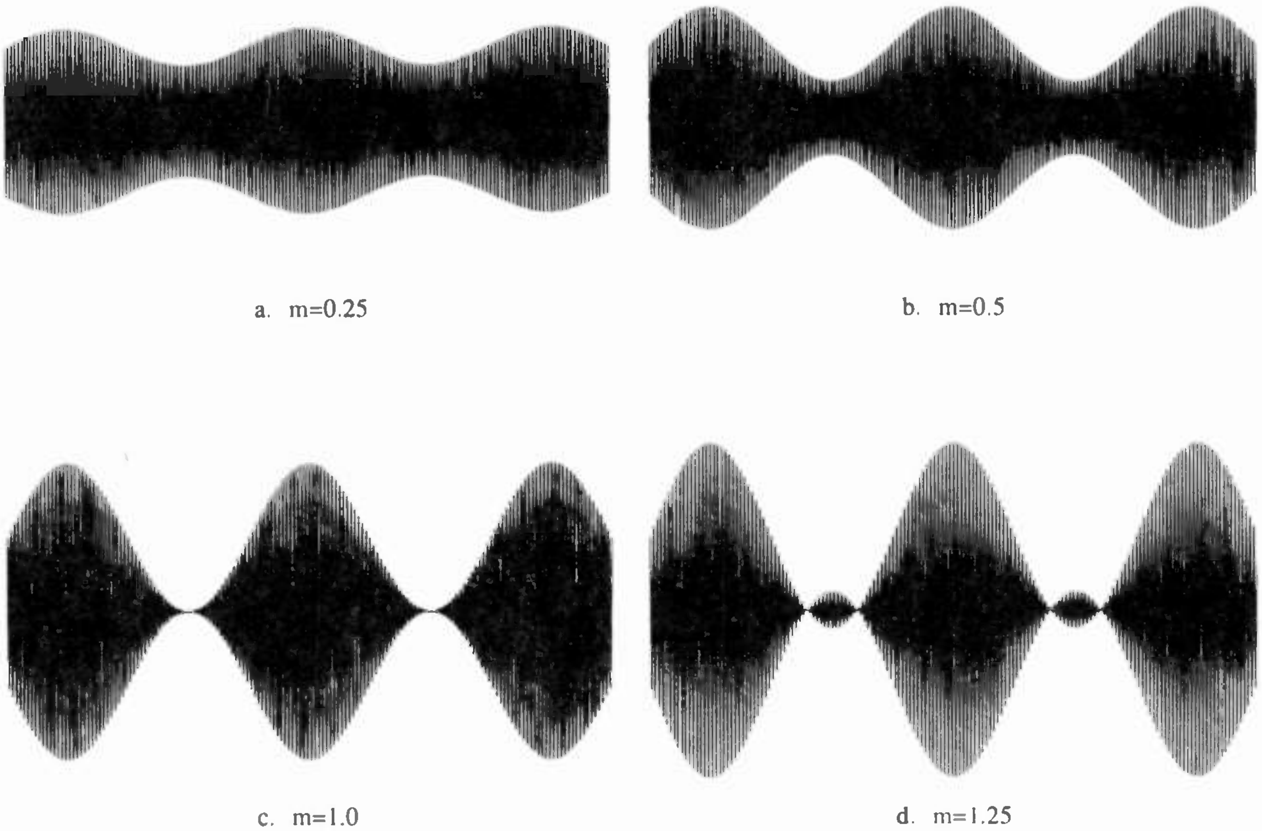


Figure 1. Waveforms of AM Carriers for Various Modulation Indices.

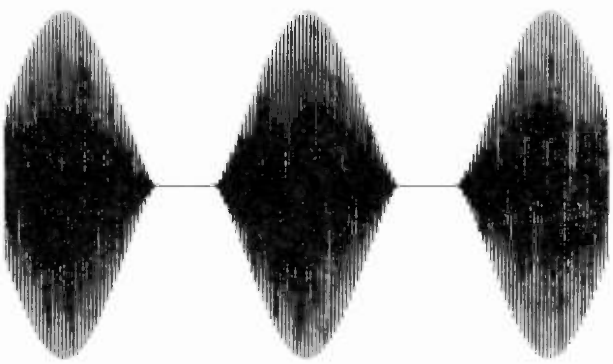


Figure 2. Overmodulation in a Typical AM Transmitter.

products or other processing artifacts reaching the transmitter input will be converted to splatter energy at the transmitter output.

V. AM Demodulation

The two conventional methods used to demodulate AM signals are envelope and synchronous detection. Envelope detection relies on the property that the envelope of an AM signal is identical, under certain conditions, to the waveform of the modulating audio.[3] An envelope detector merely rectifies and filters the AM signal to re-create the original modulating waveform, a technique that works properly only if the envelope remains undistorted during transmission. Unfortunately, imperfections between the transmitter and the receiver, some of which are discussed in Section VIII of this report, can cause envelope distortion. The resulting behavior of the AM signal during detection is important to the discussion of overmodulation.

Synchronous detection is not yet common in commercial AM receivers, but has several desirable properties that should make it much more common in the future. A synchronous detector (sometimes known as a product detector) merely repeats the multiplication process that was used to modulate the carrier. The AM signal is multiplied by a sine wave of the same frequency and phase as the transmitted carrier. It can be shown mathematically that this process, together with some simple filtering, results in the re-creation of the original modulating audio signal.[4]

A synchronous detector is not sensitive to envelope distortion. Amplitude and phase errors in the transmission path affect the received frequency response but do not cause an increase in distortion. This property, and its value in monitoring modulation, is discussed later in more detail.

It should be noted that synchronous detection is not without its practical limitations. In order to multiply the AM signal by a sine wave of the same frequency and phase as the transmitted carrier, the carrier must somehow be recovered in the receiver. If, due to such phenomena as skywave fading or co-channel interference, the carrier cannot be properly reconstructed, the quality of reception will be degraded.

VI. Theoretical Amplitude Modulation Analysis

A. Fast Fourier Transform Techniques

Modulation analysis has been greatly enhanced by computer technology. Fast Fourier Transform (FFT) techniques now make it possible to quickly and easily perform conversions between the time domain and the frequency domain, without the necessity for extensive measurements.[5] For example, a digital representation of a modulated carrier can be generated, based on a desired modulation condition. This waveform can then be converted with a forward FFT program to the frequency domain, where the spectrum can be analyzed. The spectrum can be manipulated to simulate the effect of an electrical network such as a directional antenna, and then can be reconverted with an inverse FFT program to the time domain where the envelope can be inspected for distortion. This distortion can be quantified by digitally simulating an envelope detector to produce the time domain representation of the detected audio waveform, transforming again to the frequency domain, and calculating the percentage of distortion from the levels of harmonic components.

This technique has been used herein to theoretically analyze the modulation effects of several components in the broadcast system: clippers, transmitters, directional antennas, and receivers. These effects will be discussed throughout this report.[6]

B. FFT Modulation Analysis

Figure 3 shows the results of FFT analysis of single-tone modulation under various conditions of level and clipping. The following conditions were analyzed:[7]

a. 50% modulation
b. 100% modulation
c. 100% + 3 dB (141%), no clipping or carrier pinchoff (multiplier-type overmodulation)
d. 100% + 1 dB (112%), carrier pinchoff only
e. 100% + 2 dB (126%), carrier pinchoff only

---

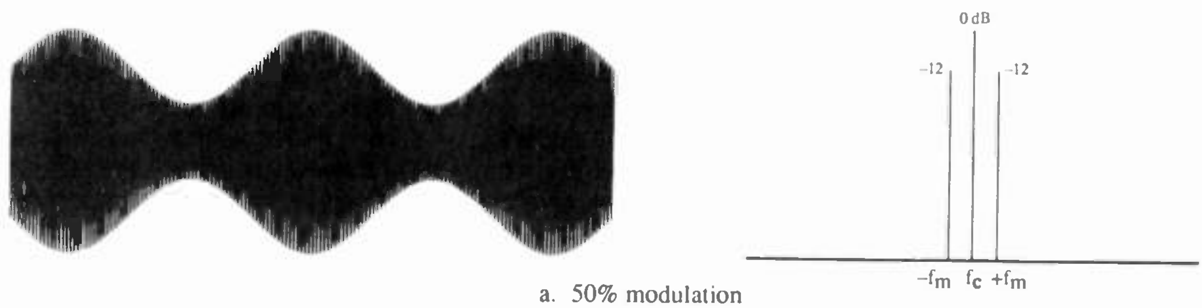
3 Figure 1 demonstrates that, up to 100% modulation, the envelope of the signal is the modulating sine wave.

4 See Appendix A for a mathematical analysis of synchronous detection.

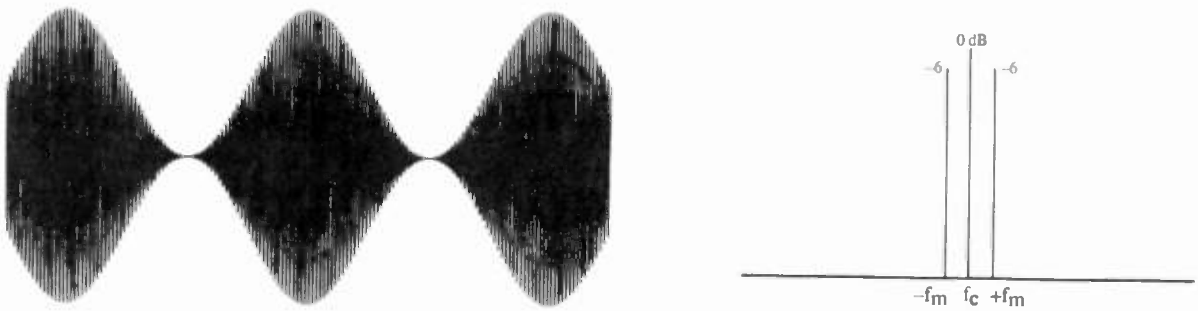
5 The Fourier Transform is a mathematical technique to convert between the time-domain representation, or waveform, of a signal, and its frequency-domain representation, or spectrum. Before the advent of computers, the use of Fourier Transforms was time consuming and limited to certain mathematical functions such as sine waves. A computer algorithm known as the Fast Fourier Transform speeds the computation of the Fourier Transform and operates on essentially any waveform or spectrum. The forward FFT converts a waveform to its spectrum, while the inverse FFT converts a spectrum to its associated waveform.

6 See Appendix B for a more complete description of FFT parameters.

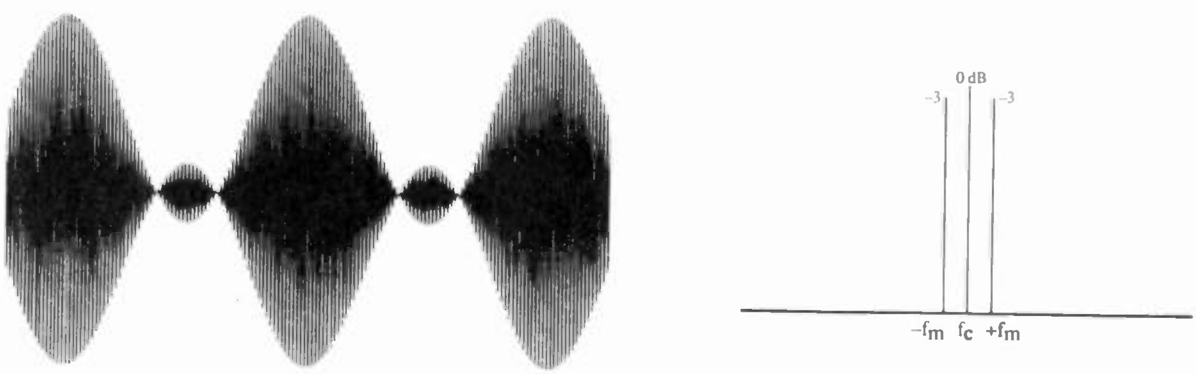
7 Most AM transmitters, whose audio stages are AC-coupled, experience a DC level shift under conditions of asymmetrical clipping that affects the peak modulation level. For example, increasing the audio level by 3 dB above 100% modulation would theoretically result in a positive peak modulation of 141%. If the negative peak were clipped to avoid negative overmodulation, an AC-coupled transmitter would shift the DC level of the audio, reducing the positive peak modulation to just over 100%. The waveforms analyzed here were analytically generated and had no DC level shift under any modulation conditions. Therefore, the FFT spectra shown here will differ slightly from those of most physical AM transmitters.



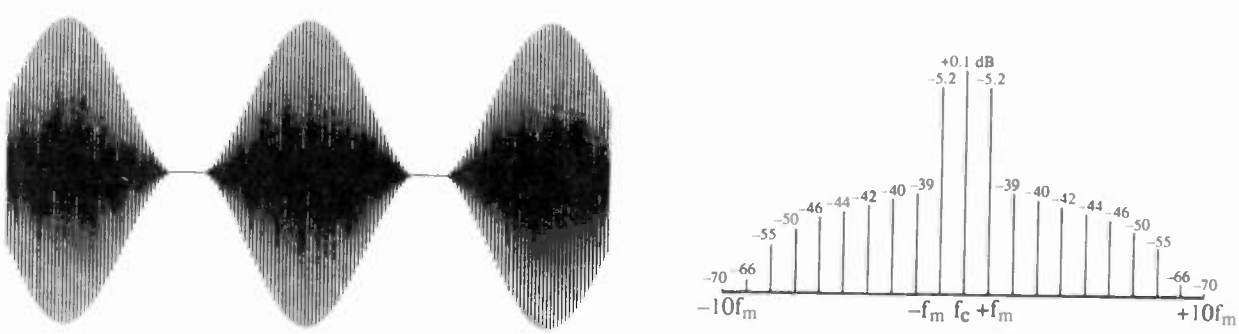
a. 50% modulation



b. 100% modulation

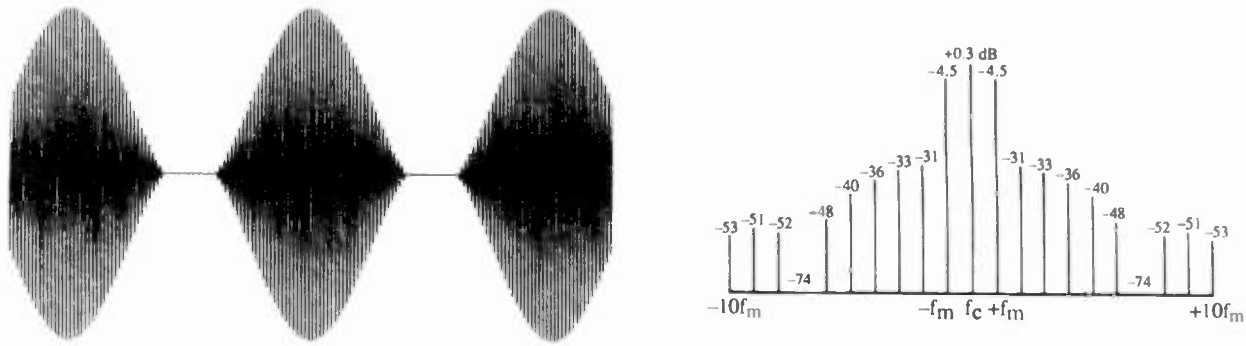


c. 100% + 3 dB (141%), multiplier-type overmodulation

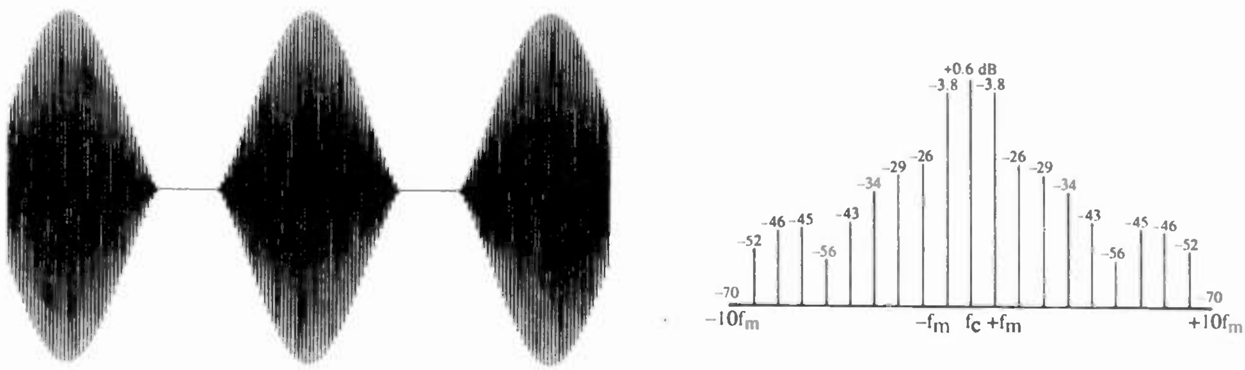


d. 100% + 1 dB (112%), carrier pinchoff only

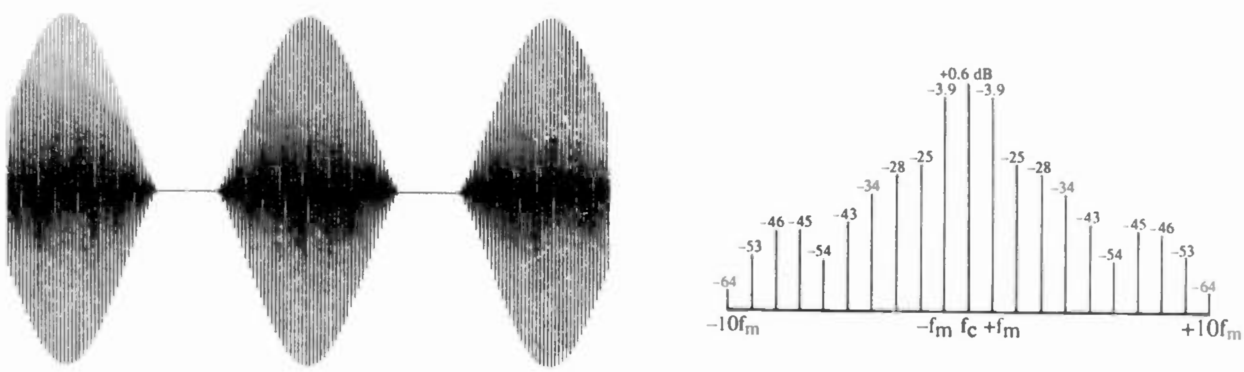
Figure 3. FFT Modulation Analysis - Waveforms and Spectra.



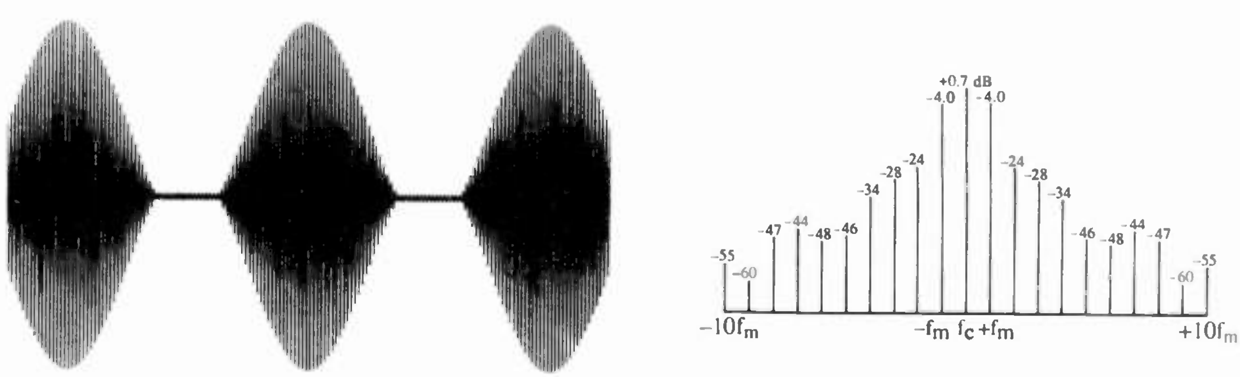
e. 100% + 2 dB (126%), carrier pinchoff only



f. 100% + 3 dB (141%), carrier pinchoff only

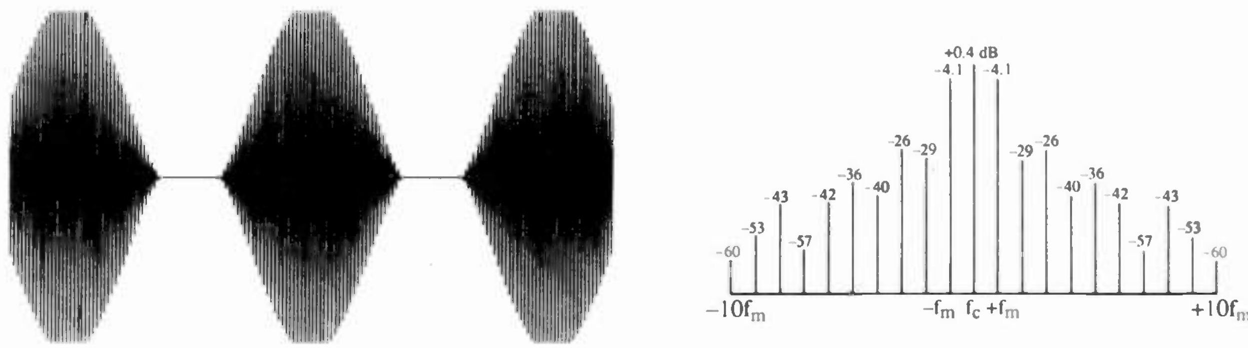


g. 100% + 3 dB, –99% audio clip

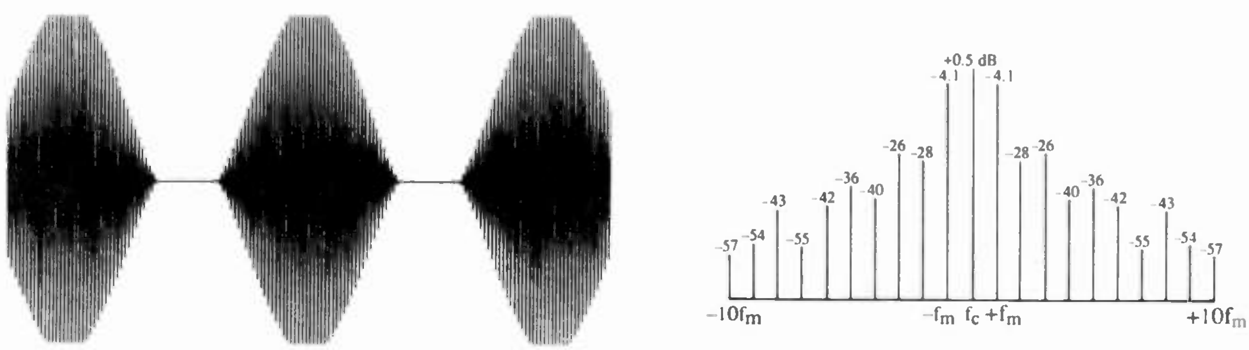


h. 100% + 3 dB, –95% audio clip

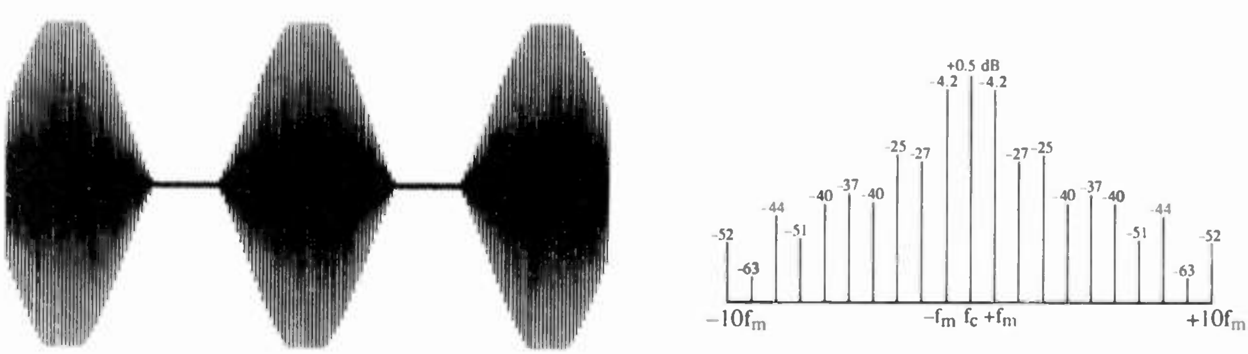
Figure 3 (cont.). FFT Modulation Analysis - Waveforms and Spectra.



j. 100% + 3 dB, carrier pinchoff, +125% audio clip



k. 100% + 3 dB, –99% audio clip, +125% audio clip



l. 100% + 3 dB, –95% audio clip, +125% audio clip

Figure 3 (cont.). FFT Modulation Analysis - Waveforms and Spectra.

f. 100% + 3 dB (141%), carrier pinchoff only
g. 100% + 3 dB, -99% audio clip
h. 100% + 3 dB, -95% audio clip
j. 100% + 3 dB, carrier pinchoff, +125% audio clip
k. 100% + 3 dB, -99% audio clip, +125% audio clip
l. 100% + 3 dB, -95% audio clip, +125% audio clip.

These figures demonstrate some important basic properties of modulation and splatter. First, the intermittent disappearance of the carrier, which is believed by many broadcast engineers to be the cause of splatter, actually has very little to do with it. Multiplier-type overmodulation, shown in Figure 3c, results in the carrier often going to zero but does not result in the generation of any new frequency components; conversely, Figures 3g, 3h, 3k, and 3l show that splatter can occur even though the carrier does not disappear. Second, it is not "sharp edges" of the modulation, per se, that produce splatter; again, the zero crossings of multiplier-type overmodulation are sharp but cause no splatter.

Most important, it is apparent from Figure 3 that negative peak clipping is the major cause of splatter and that, theoretically, it matters little whether the clipping occurs as carrier pinchoff in the transmitter or as clipping in the audio prior to modulation. Comparing the spectra of 3 dB overmodulation in Figures 3f through 3l, there is almost no difference between carrier pinchoff and audio clipping. The conditions of carrier pinchoff, –99% audio clipping, and –95% audio clipping, produce spectral components which are stronger than –35 dB out to the fourth harmonic of the modulating frequency, and produce additional significant components even farther out.

When +125% audio clipping is added to an overmodulated signal, as in Figure 3j, or to a negatively clipped signal, as in Figures 3k and 3l, the effect is less significant. While some spectral components are increased if positive clipping is added, others are actually reduced due to the phase relationships between the components produced by positive and negative clipping. Also, the larger components produced when negative peaks are clipped from 141% down to 100% masks the smaller components produced when positive peaks are clipped from 141% down to only 125%. The net additional splatter energy produced by +125% clipping is small.

It can be concluded from this data that the existing +125% FCC modulation limit is counterproductive. In practice, a station adjusts its modulation level to the –100% limit, then clips positive peaks to comply with the +125% limit. Contrary to the original FCC intention, such clipping has only a negative effect on occupied bandwidth. Although the effect is a small one, occupied bandwidth would benefit by removing the artificial restriction on positive peaks.

VII. Transmitter Modulation Measurements

To confirm the FFT results, and to analyze the spectral characteristics of broadcast program material, spectrum analyzer measurements were made using the facilities of nearby AM Station KOFY, San Mateo, California. The station is a one-kilowatt non-directional daytimer with two alternate transmitters, a new all-solid-state model and a much older plate-modulated model. The test configuration is shown in Figure 4. Figure 5 shows how to interpret the spectrum analyzer photographs.

Initial tests were made using both the dummy load and the antenna. Due to the wide bandwidth of the single tower, its test results were virtually identical to those of the dummy load so, as a convenience, the dummy load was used for subsequent tests. More complex antenna systems would produce different amounts of sideband energy than these test results show; most will have a narrower bandwidth than does a single tower so will produce less sideband energy. However, at certain azimuths from a directional antenna the sideband energy may be greater, relative to the carrier, than it would be in a wideband system.[8]

A. Sine Wave Measurements

Figure 6 shows the spectrum analyzer displays for the solid-state transmitter under the following modulation conditions:

a. Unmodulated carrier
b. 50% modulation, 1 kHz tone
c. 100% modulation, 1 kHz tone
d. 100% + 3 dB, 1 kHz tone, carrier pinchoff only
e. 100% + 3 dB, 1 kHz tone, -95% audio clip[9]
f. 100% modulation, 10 kHz tone
g. 100% + 3 dB, 10 kHz tone, carrier pinchoff only
h. 100% + 3 dB, 10 kHz tone, -95% audio clip.

By comparing these measured results with the corresponding figures showing the calculated FFT spectra, it can be seen that the methods correlate well. The sideband amplitudes are similar, and show the equivalence of carrier pinchoff and audio clipping for lower modulating frequencies.

For the 10 kHz modulating frequency, shown in Figures 6f through 6h, audio clipping generates fewer sidebands far from the carrier. This occurs because, although FFT analysis showed that audio and carrier clipping have essentially identical effects, carrier clipping will produce more damaging interference in practice because of limited transmitter high-frequency response. A carrier that is overmodulated with a 10 kHz audio signal is clipped at a 10 kHz rate, generating spurious components at 20 kHz, 30 kHz, 40 kHz, etc. Their amplitudes are limited only by the bandwidth of the antenna. If the same audio signal is clipped prior to reaching the transmitter,

8 For example, a directional antenna system is usually tuned for minimum radiated field at carrier frequency in a pattern minimum. The null will probably not be as deep at sideband frequencies in a narrowband system, so sideband energy will be enhanced relative to the carrier.

9 Because clipped sine waves were DC-level-shifted by the AC-coupled transmitter, +125% sine wave modulation was never achieved.

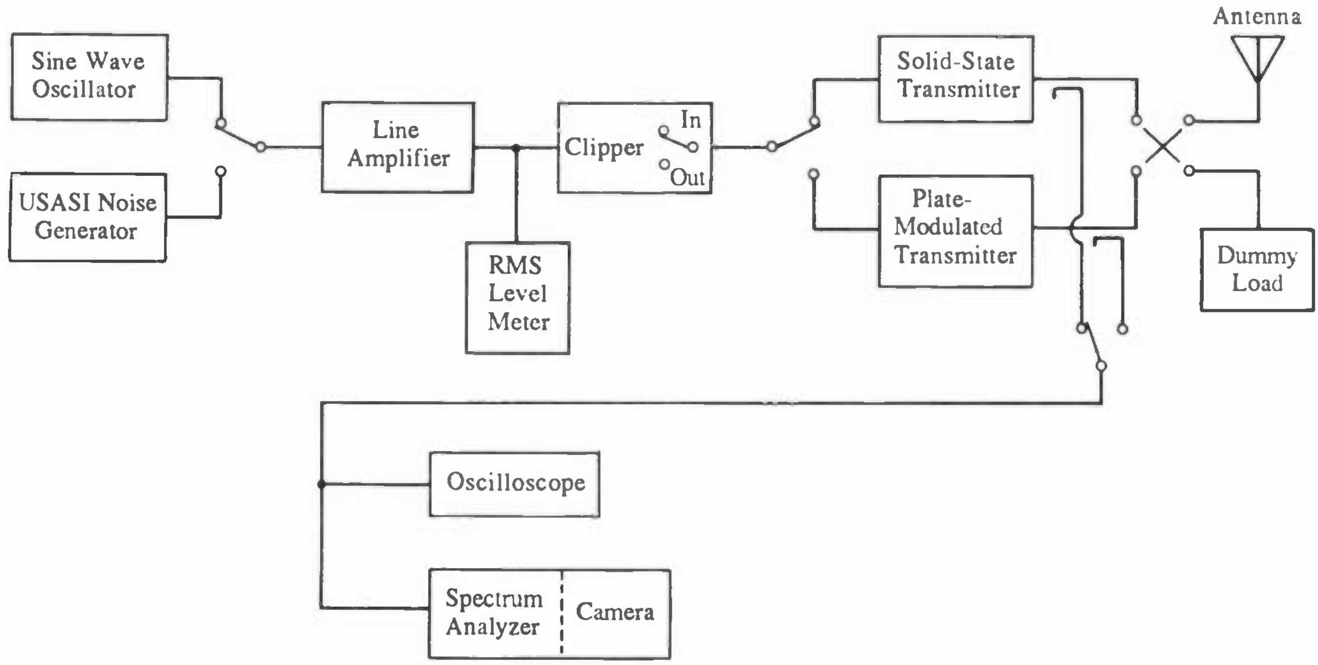


Figure 4. Transmitter Modulation Measurements - Equipment Configuration.

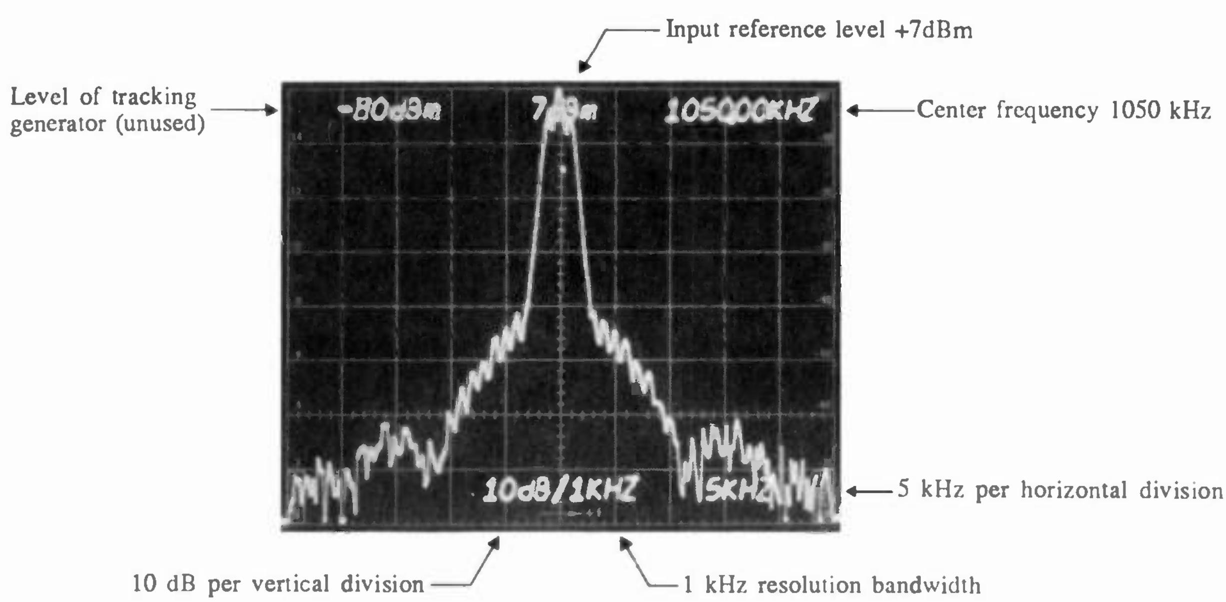
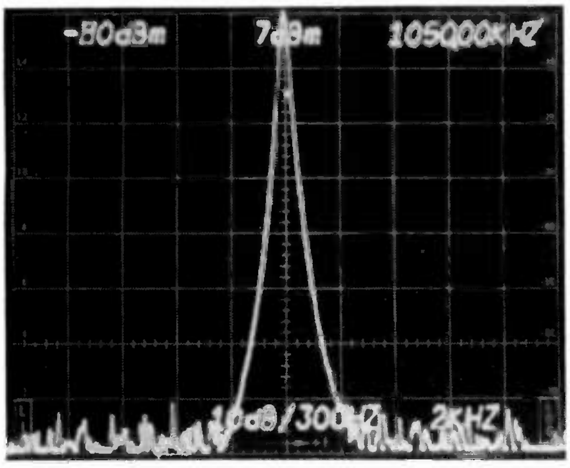
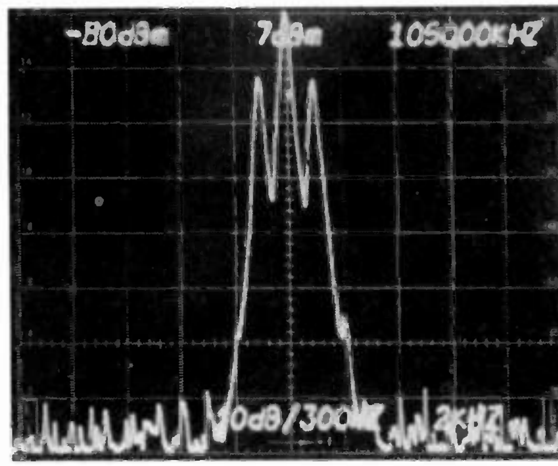


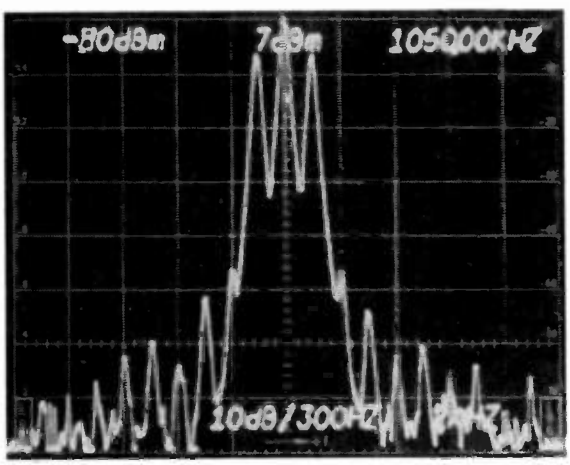
Figure 5. Interpretation of Spectrum Analyzer Photographs.



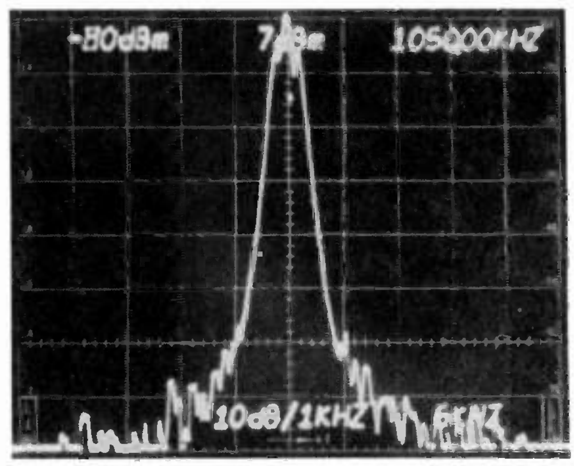
a. Unmodulated carrier



b. 50% modulation, 1 kHz tone

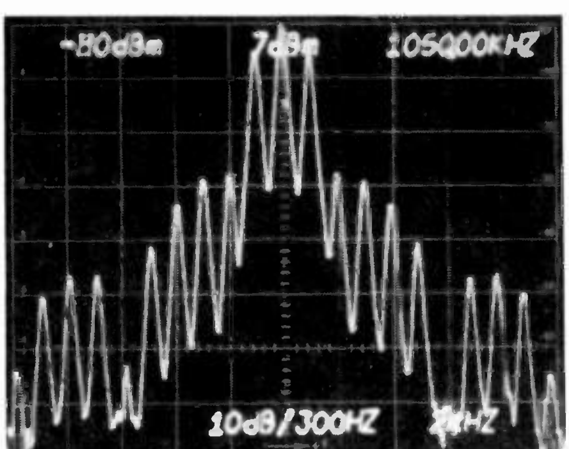


2 kHz/division

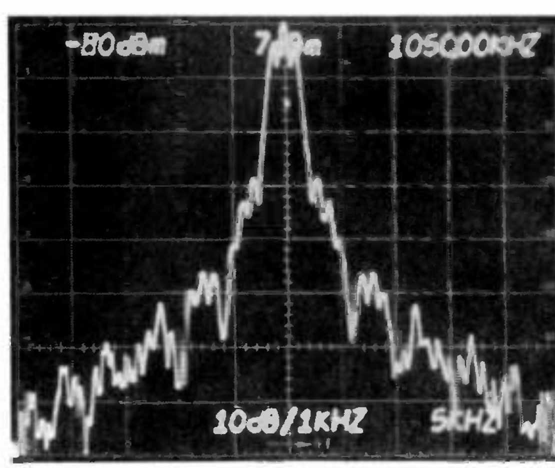


5 kHz/division

c. 100% modulation, 1 kHz tone



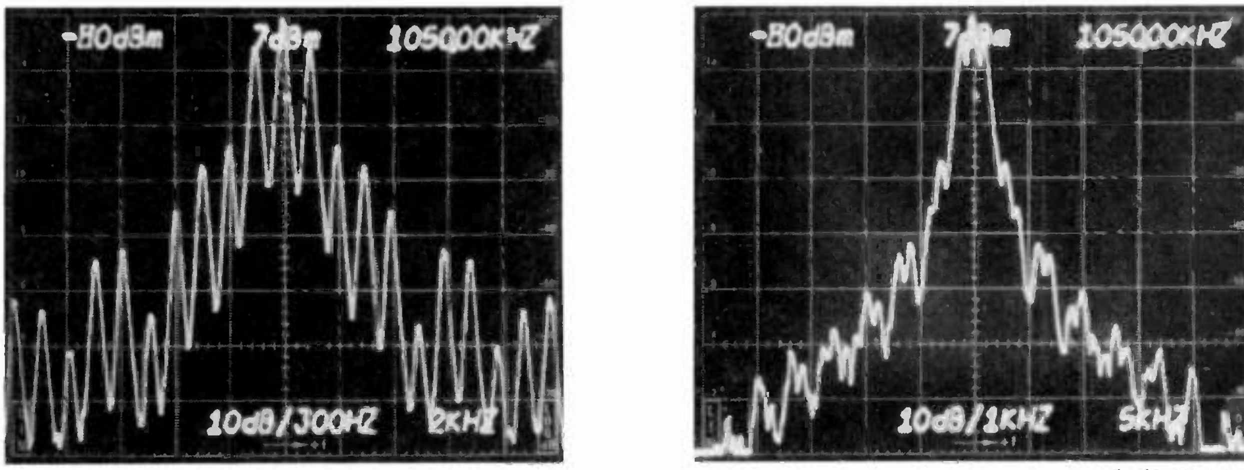
2 kHz/division



5 kHz/division

d. 100% + 3 dB, 1 kHz tone, carrier pinchoff only

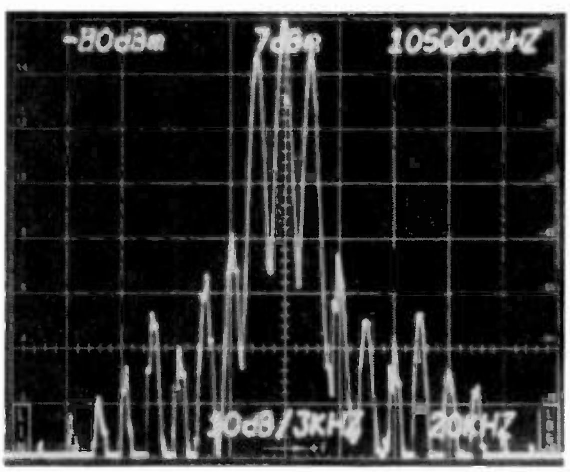
Figure 6. Spectrum Analyzer Displays, Solid-State Transmitter.



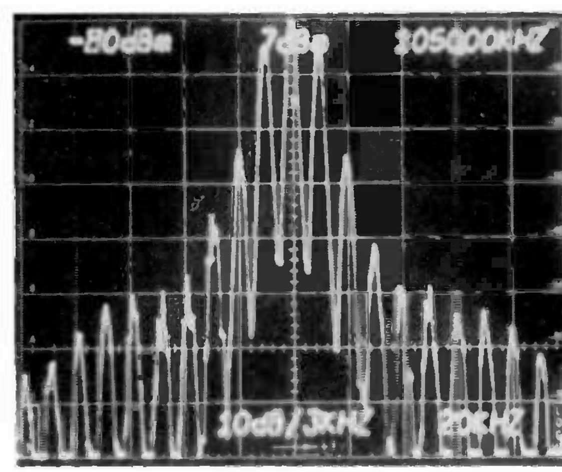
2 kHz/division

5 kHz/division

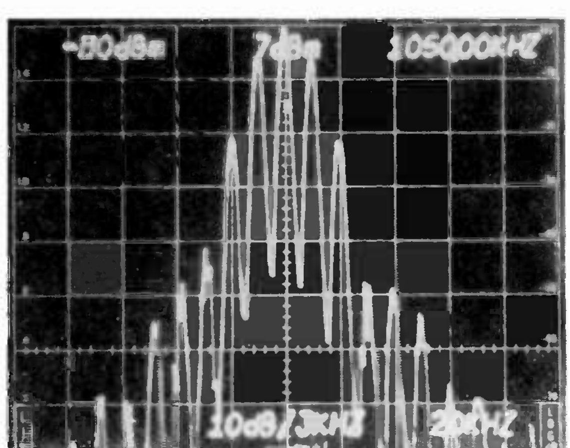
e. 100% + 3 dB, 1 kHz tone, –95% audio clip



f. 100% modulation, 10 kHz tone



g. 100% + 3 dB, 10 kHz tone, carrier pinchoff only



h. 100% + 3 dB, 10 kHz tone, –95% audio clip

Figure 6 (cont.). Spectrum Analyzer Displays, Solid-State Transmitter.

much of its harmonic energy will be rolled off in the transmitter and will not reach the antenna.

Modern transmitters have excellent high-frequency response so, despite the roll-off, audio clipping can produce significant spurious energy at frequencies as much as 40 kHz from the carrier; significant energy from carrier clipping extends even farther. The figures show that, with only a 1 kHz modulating frequency, splatter components up to 20 dB above the noise level occur 17 kHz from the carrier. 10 kHz overmodulation produces such splatter components 80 kHz away.

These figures also strikingly demonstrate that *meeting the FCC occupied bandwidth limitations is no guarantee of a clean signal*. Section 73.44 of the FCC Rules requires emissions from an AM station to be 25 dB below carrier amplitude between 15 kHz and 30 kHz from the carrier, 35 dB below carrier between 30 kHz and 75 kHz from the carrier, and 67-80 dB below carrier (depending on station power) more than 75 kHz from the carrier. Even the worst-case scenario of 3 dB overmodulation at 10 kHz, a condition that would never occur in practice, meets these limits up to 75 kHz.

Figure 7 shows similar spectrum analyzer displays for the plate-modulated transmitter:

a. 100% modulation, 1 kHz tone
b. 100% + 3 dB, 1 kHz tone, carrier pinchoff only
c. 100% + 3 dB, 1 kHz tone, -95% audio clip
d. 100% modulation, 10 kHz tone
e. 100% + 3 dB, 10 kHz tone, carrier pinchoff only
f. 100% + 3 dB, 10 kHz tone, -95% audio clip.

This transmitter had significantly more distortion than the newer solid-state version, and showed some differences in levels of specific sidebands, but its splatter characteristics were not significantly different. Nothing in the test results indicated that old transmitters generated more splatter than did newer ones.[10]

B. Noise Measurements

Sine waves are useful analytical tools with which to gain a basic understanding of overmodulation, but any accurate description of the effects of overmodulation on actual broadcast stations must focus on program material. For these tests, program material was simulated by "USASI" (United States of America Standards Institute) noise. This is a type of weighted noise, developed for sound level meters, with a spectral characteristic that was empirically designed to be similar to average programming. It consists of white noise that has been filtered by a 100 Hz, 6 dB per octave high-pass network and a 320 Hz, 6 dB per octave low-pass network. It has recently been concluded that USASI noise continues to be a close approximation to current music program material.[11] The source for USASI noise in these tests was the General Radio Model 1382 Random Noise Generator. Figure 8, taken from the unit's instruction manual, shows the spectrum of USASI noise compared to white and pink noise.

It was desired to simulate the modulation density of a typical AM radio station. By analyzing the peak and average level characteristics of a number of stations, it has been found that the lowest peak-to-RMS ratio, corresponding to the most heavily processed station, is approximately 6 dB to 9 dB, depending on the integration time of the RMS detector.[12] 9 dB was assumed as an appropriate value for these tests. Broadcast stations have a peak modulation level of approximately 100%; a sine wave has a peak-to-RMS ratio of 3 dB. Therefore, a reference 0 dB noise level for these tests was defined to be 6 dB below the RMS sine wave level needed to modulate the transmitter to 100%.

When applied to the solid-state transmitter, the 0 dB USASI noise produced frequent negative modulation peaks of 100%, with positive peaks exceeding 110%. As the level of the noise was increased by up to 6 dB, the amount of negative overmodulation increased and the positive modulation exceeded 150%. To evaluate the different effects that carrier pinchoff and audio clipping had on occupied bandwidth, the audio clipper was adjusted for –95% and +125% modulation, and the spectrum was compared at several noise levels with and without audio clipping.

The noise was not preemphasized for the tests. The amount of high-frequency boost used by AM stations varies widely, so any particular choice for a preemphasis curve would have been arbitrary and applicable only to some stations. USASI noise itself is believed to be representative of the spectral characteristics of actual program material, so the actual spectrum expected from a particular radio station can be extrapolated by adding the station's preemphasis curve to the curve shown on the spectrum analyzer.

The noise was also not band-limited for the tests. Although the noise amplitude decreases by 6 dB per octave above 320 Hz, there is still some energy present above 20 kHz, differing from normal programming in which little spectral content exists above 15 kHz. The effect of this difference on the test results is discussed in Section VII–C, below.

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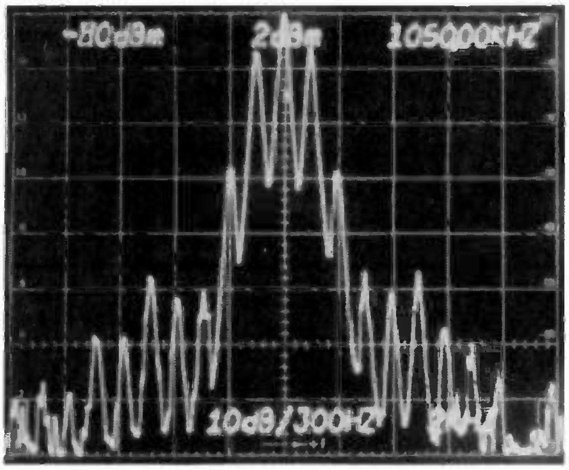
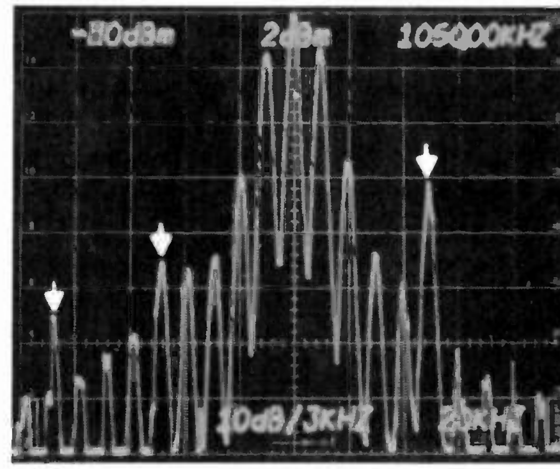
10 Transmitter Transient Distortion (TTD) has been suggested as a possible cause, in some transmitters, of spurious emissions that are not noticed in steady state measurements. This phenomenon was not investigated during this project. For a discussion of TTD, *see AM Technical Improvement Report*, NAB AM Improvement Subcommittee, October 1984 at 44.

11 *See* Payne, Christopher P., *The Characterization of Amplitude Response in Audio Systems Employing Program Dependent Variable Equalization*, March 1986, submitted to the National Radio Systems Committee and available from the author at (202) 862-1549.

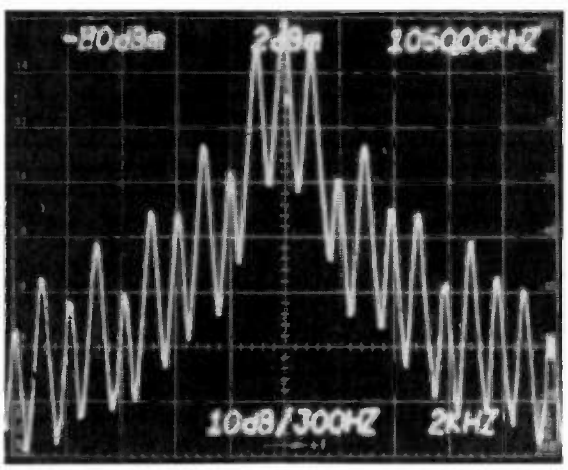
12 Personal communications with Christopher Payne (id.) and Robert Orban, Orban Associates, Inc., (415) 957–1063.



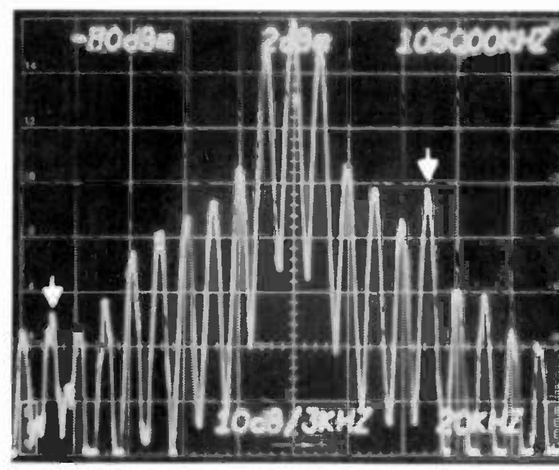
a. 100% modulation, 1 kHz tone



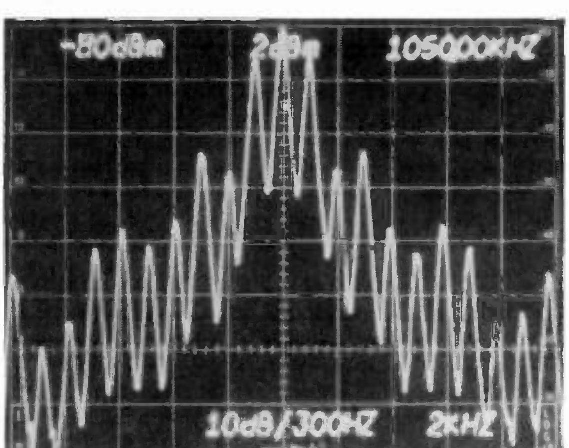
d. 100% modulation, 10 kHz tone (note: 960 kHz, 1000 kHz, and 1100 kHz components are other stations)



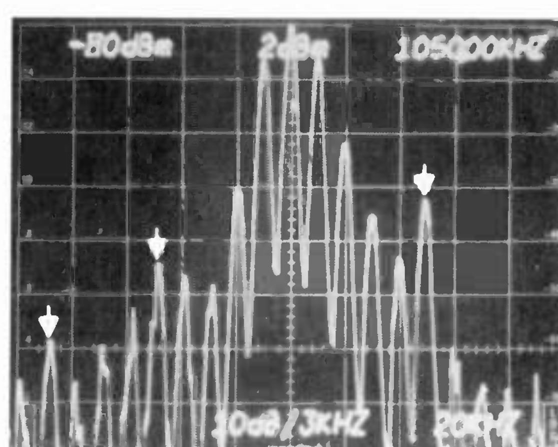
b. 100% + 3 dB, 1 kHz tone, carrier pinchoff only



e. 100% + 3 dB, 10 kHz tone, carrier pinchoff only (960 kHz and 1100 kHz components are other stations)



c. 100% + 3 dB, 1 kHz tone, -95% audio clip



f. 100% + 3 dB, 10 kHz tone, -95% audio clip (960 kHz, 1000 kHz, and 1100 kHz components are other stations)

Figure 7. Spectrum Analyzer Displays, Plate-Modulated Transmitter.

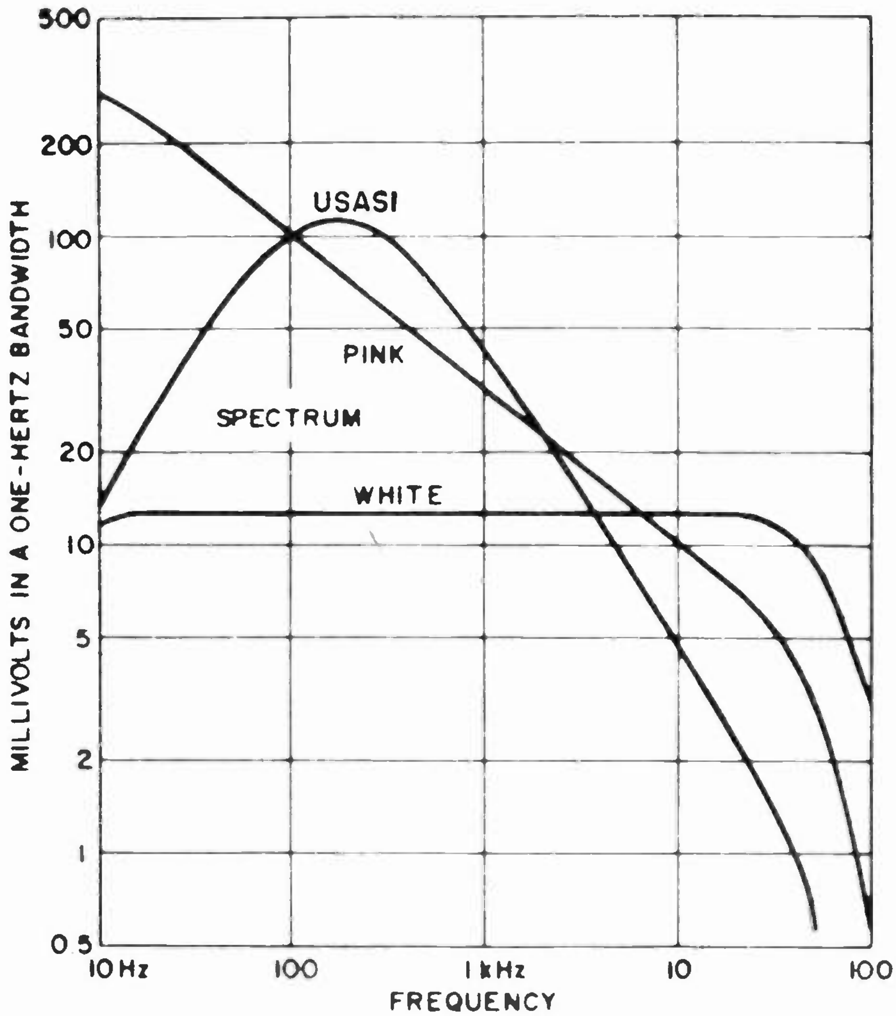


Figure 8. Spectra of White, Pink, and USASI Noise.

Figure 9 shows the spectra of the measured noise-modulation conditions. The spectrum analyzer was adjusted for a display of ±50 kHz, with a resolution bandwidth of 3 kHz. The analyzer was left in the MAX HOLD mode for approximately two minutes, after which virtually no further change in the curve was noted. The reference amplitude at the top of the display was adjusted to be the unmodulated carrier amplitude.[13] Spectrum measurements were made for noise levels of –3 dB, 0 dB, +3 dB, and +6 dB. Since the 0 dB noise reference level was chosen as equivalent to a highly processed station, +6 dB is a true "worst case." It is a far greater average program level than could possibly exist in practice, equal to the level of a constant 100% sine wave.

The spectra shown in Figure 9 are quite surprising. A USASI input noise level of –3 dB (Figure 9a) results in peak modulation levels of less than 100%. A noise level of +6 dB (Figure 9d) results in almost continuous negative overmodulation and positive modulation levels exceeding 150%, and requires heavy audio clipping if overmodulation is to be avoided. Yet the RF spectrum shape is identical at these two extremes, for both carrier clipping and audio clipping conditions. For each 3 dB increase in noise level, the spectrum increases by approximately 3 dB at all frequencies. One might have expected the sidebands of the heavily clipped +6 dB signal to be significantly greater than 9 dB above the level of the lightly modulated –3 dB signal, due to the generation of distortion components at the higher levels. This shows that the spurious components produced by either overmodulation or clipping are masked by the higher frequencies already present in the modulating signal. There is little penalty in occupied bandwidth due to clipping or overmodulation under these modulation conditions.[14]

C. Band-Limited Noise

The use of a low-pass filter on USASI noise, to more closely simulate the spectral characteristics of broadcast program material, would affect the results of these tests. Overmodulation and clipping has such little effect on the measured USASI noise spectra because the spurious components are masked by the high-frequency components already present. Sharp low-pass filtering of the input USASI noise would remove these masking components, making the spurious components more significant.

In particular, Orban has measured the output power spectrum of an Orban 9100A processor, which includes a sharp 12 kHz low-pass filter.[15] Harmonic components caused by the processor's clipper generate a long spectrum "tail" beyond 14 kHz. These components are approximately 45 dB down at 15 kHz, dropping to 80 dB down by 45 kHz. Accordingly, the spurious products of an actual clipped or overmodulated broadcast signal, although still within FCC limits, would be likely to cause greater adjacent-channel interference than would a clean signal without such products. The better the processor is at controlling spurious high-frequencies, the more noticeable will be any splatter caused by post-processor clipping or overmodulation.

D. Occupied Bandwidth Analysis of Noise Modulation

All of the measured spectra in Figures 9a through 9d meet FCC occupied bandwidth criteria, although 10 dB of preemphasis at 15 kHz would cause the -25 dB FCC limit to be exceeded over a narrow band at the 0 dB noise level or higher. This result substantiates our observation on the non-constraining nature of the FCC limits.

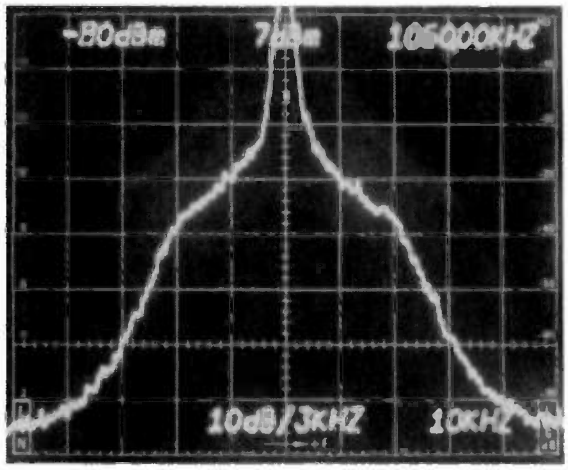
Although only modest audio filtering would be required to meet FCC occupied bandwidth requirements even under these most egregious modulation conditions, it is good engineering practice to seek lower sideband levels at frequencies far from the carrier. As mentioned above, USASI noise contains more energy above 20 kHz than does program audio, so the test spectra appear worse than they would under program conditions. However, to minimize interference to second- and higher-adjacent-channel stations, protective filtering is advised even with program audio. Most popular audio processors contain a low-pass filter with a sharp cutoff characteristic above 12 kHz, so as to be at least 25 dB down at 15 kHz. The National Radio Systems Committee (NRSC) is presently discussing the potential advantages of lower cutoff frequencies in certain allocation circumstances.

It can be concluded from Figure 9 and the discussions in this section that the elimination of overmodulation would have only a modest effect on the present character of the AM band. While overmodulation may create some low-level spurious energy far from the carrier, the primary component of sideband energy in nearby adjacent channels is the energy in the program material. As the high-frequency content of program material has increased, as the amount of preemphasis has increased, as antenna and

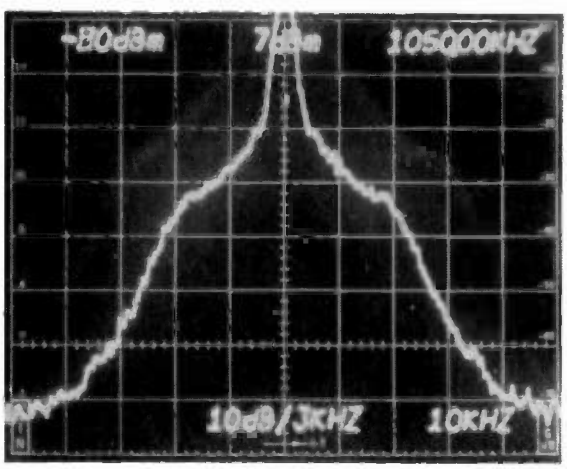
[13] The figures appear to show a carrier amplitude greater than the reference level. This occurs in the MAX HOLD mode because the maximum energy in the 3 kHz resolution bandwidth, which includes the carrier and the lower-frequency sidebands, exceeds the carrier amplitude alone.

[14] It was mentioned in Section VII-A that, for high modulating frequencies, carrier pinchoff produces more splatter than does audio clipping. This is not evident in Figure 9. Therefore, there was some concern that the non-preemphasized noise was not exciting the splatter-generating mechanisms in a realistic way, because the high-frequency components of USASI noise are below the level necessary to cause carrier clipping. However, these figures show that, even with 15 dB preemphasis, a 10 kHz noise component remains below the clipping threshold. It was concluded that non-preemphasized USASI noise was a realistic modulating waveform for this analysis.

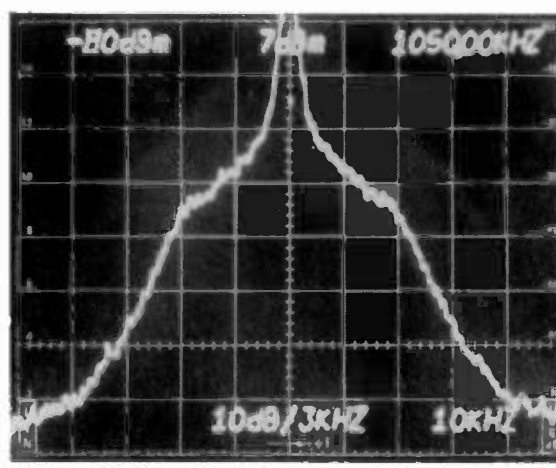
[15] *See Minutes of Subgroup on Methods and Procedures of the National Radio Systems Committee*, Attachment C (May 21, 1986), available from NAB Science and Technology, (202) 429-5346.



a. -3 dB input level (<100% peak modulation)



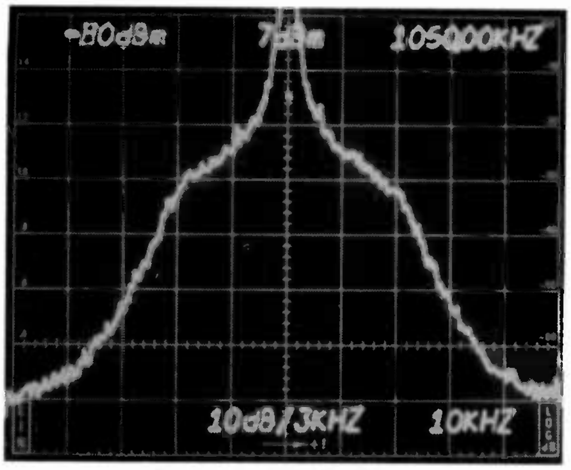
carrier pinchoff only



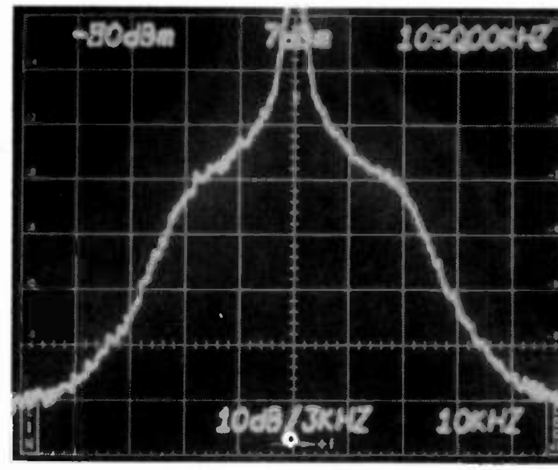
-95% audio clip (<+125% peak modulation)

b. 0 dB input level

Figure 9. Spectrum Analyzer Displays, USASI Noise Modulation.

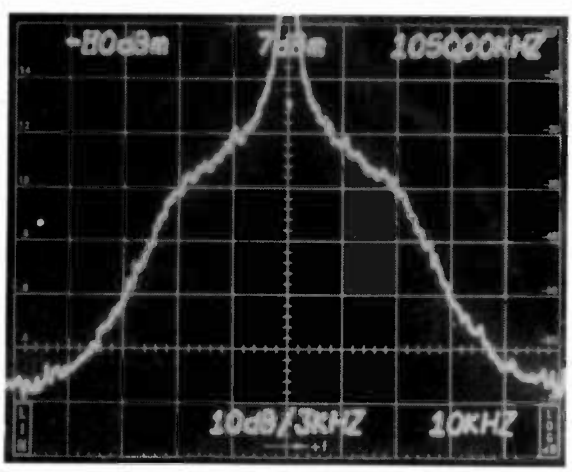


carrier pinchoff only

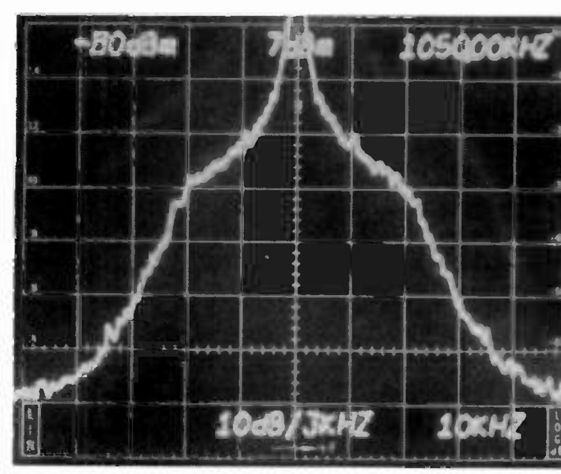


–95%, +125% audio clip

c. +3 dB input level



carrier pinchoff only



-95%, +125% audio clip

d. +6 dB input level

Figure 9 (cont.). Spectrum Analyzer Displays, USASI Noise Modulation.

transmitter bandwidths have increased, and as the number of stations has increased, the amount of interference has greatly increased and the limitations of 10 kHz channel spacing have become more apparent. Overmodulation is only a minor component. *Major reductions in splatter interference can only be achieved by reducing the high-frequency energy content of the modulating signal.*

E. DC Level Shift

Overmodulation can be exacerbated by the DC level shift in AC-coupled transmitters. Station engineers often meticulously adjust their processing to reach –98% modulation, only to find that overmodulation occurs with different program material. Even if the peak levels leaving the processor are tightly controlled, AC-coupling can shift them by several percent as the asymmetry of the program material varies. This can be enough to cause overmodulation in an otherwise well-designed audio processing system.

DC level shift is a common problem in television but is rarely addressed in radio. Several stations have modified their AM transmitters to full DC-coupled operation. Other engineers have experimented with a DC clamp that stops the final amplifier of the transmitter from cutting off the carrier; while this does not prevent DC level shift, it eliminates its negative effects. These techniques are effective but of limited use. DC-coupling can only be accomplished in some types of transmitters; plate-modulated transmitters, for example, cannot be DC-coupled. In addition, many engineers are uncomfortable with the existence of a DC path at the audio input.

Fortunately, the problem can be essentially solved, without the necessity for complete DC-coupling, by improving the low-frequency response of the transmitter. It has been suggested that a reduction of the lower 3 dB cutoff frequency to 0.1 Hz should be sufficient.[16] Transmitter engineers should develop field modification kits for such operation and include this feature in newer models.

F. Minimizing Occupied Bandwidth

As a result of the measurements and analysis of sine-wave and noise modulation, the following steps are recommended to minimize excessive occupied bandwidth:

1. All audio processing equipment should contain or be followed by an appropriate overshoot-corrected low-pass filter to minimize spectral components above the desired audio range.
2. Because high-frequency audio clipping produces less splatter than does high-frequency carrier clipping, a protective audio clipper is advised as the last device before the transmitter. Such a clipper is often contained in modern audio processors; if so, it is preferable to a separate device. If a separate device is necessary, it can be a stand-alone unit or may already be built into the transmitter input circuitry. The clipping point of a separate device should be approximately -95%, to insure that isolated high-frequency peaks passed by the audio processing system do not cause carrier pinchoff in the transmitter, and the audio output level from the processor should be just below the level at which clipping occurs. *The protection clipper must not be used to increase loudness.* New clippers should be designed to provide an indication of clipping amount so that excessive clipping can easily be recognized.
3. To minimize DC level shift, which can cause unwanted carrier clipping of audio signals having even well-controlled peak levels, stations should investigate reducing the low-frequency cutoff point of their transmitters to approximately 0.1 Hz, or converting their transmitters to DC-coupled or DC-restored operation. Transmitter engineers should develop field modification kits for such operation for older transmitters and include these features in newer models.

VIII. Amplitude and Phase Errors

A. Envelope Distortion

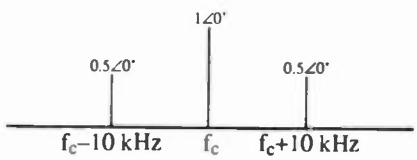
The limited-bandwidth circuitry in a transmitter output network or antenna system can change the amplitude and phase of the spectral components of an amplitude-modulated signal. Because these amplitude and phase errors can affect the apparent modulation level and can cause distortion in receivers, it is easy to mistakenly conclude that antenna systems can cause transmitter-type overmodulation and splatter. This is not correct. Transmitter-type overmodulation is a non-linear process that does not occur in linear electrical networks made up of only inductors, capacitors, and radiation resistance. These elements can affect the amplitude and phase of a spectral component, but they cannot create new frequencies.

Although RF networks do not cause splatter, their effects on the signal can be significant. Unless understood, these effects can lead station engineers to make changes in their audio processing that distort the signal for most listeners. It is important that station engineers understand the actual mechanism by which these effects occur so they can properly make compensating adjustments or can recognize the need for design changes.

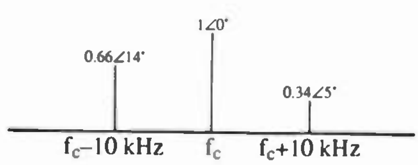
FFT techniques can be used to graphically demonstrate these effects. Figure 10 shows the effect that a limited-bandwidth antenna can have on the spectrum of an AM signal.[17] The first figure shows the theoretical three-component spectrum when the carrier is 100%-modulated with a 10 kHz tone; the sideband components are 6 dB

[16] Personal communication with Robert Orban (id.).

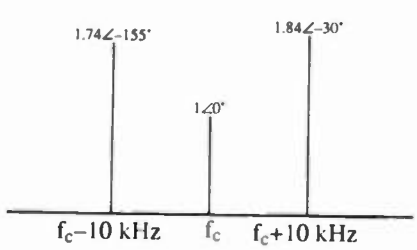
[17] These figures were derived from the measured tower currents and phases of an actual two-tower directional antenna.



a. Theoretical Spectrum

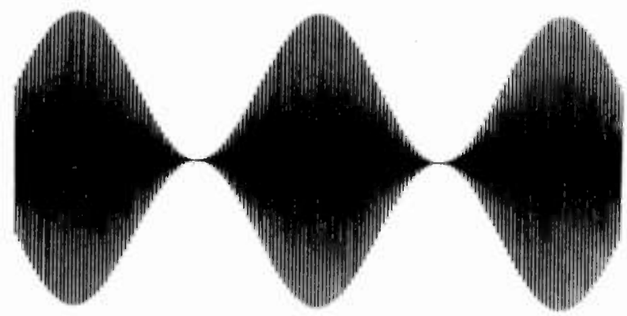


b. Spectrum in Major Lobe

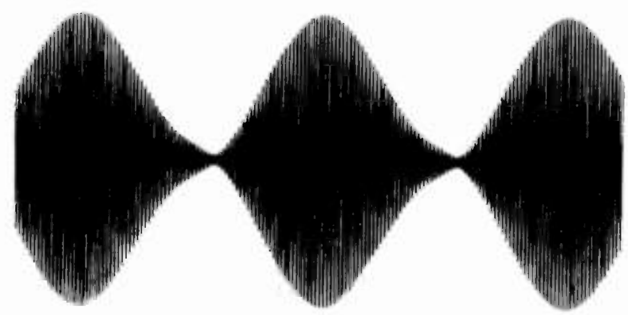


c. Spectrum in Pattern Minimum

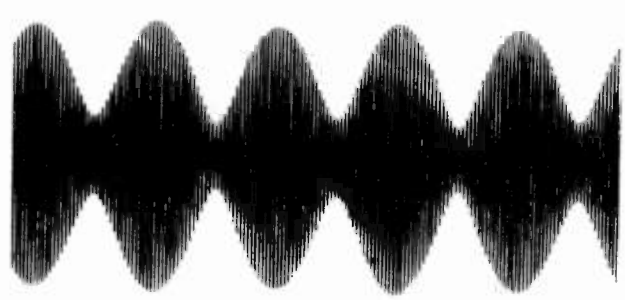
Figure 10. Spectrum Effects of Limited-Bandwidth Antennas.



a. Theoretical Waveform



b. Waveform in Major Lobe



c. Waveform in Pattern Minimum

Figure 11. Waveform Effects of Limited-Bandwidth Antennas.

below the carrier and the three components are in phase. The second and third figures show the actual spectra that would be found in the major lobe and in the pattern minimum, respectively. The amplitudes and phases of the sideband components have been normalized for a carrier component of $1\angle 0°$; they show significant differences from the theoretical spectrum.

To show what effect these amplitude and phase distortions have on the received signal, these spectra were converted to their time-domain waveforms with an inverse Fast Fourier Transform. The waveforms, shown in Figure 11, are those that would be detected by an AM receiver in the field. The amplitude and phase distortions in the frequency domain have been converted to envelope distortions in the time domain. In the major lobe, the envelope still approaches 100% modulation but some distortion is visible. In the direction of the array minimum, the fundamental 10 kHz frequency of the modulating waveform is not even discernable; only a distorted second harmonic of reduced modulation percentage is apparent.

This envelope distortion was quantified by digitally synthesizing an envelope detector and using the detected waveforms as inputs to a direct FFT. The output spectrum of each transform, containing the 10 kHz audio fundamental together with the various harmonic distortion components, was converted to a Total Harmonic Distortion (THD) figure. THD in the major lobe is approximately 5%, while in the pattern minimum it is over 1500%![18] Although there were no undesired frequency components created in the transmitted RF spectrum by the amplitude and phase errors in the antenna system, there were many such components created in the envelope-detected audio spectrum. While these distortion components can be harmful to the station's audio quality, and may sound similar to RF splatter, they do not cause interference to other stations.

B. Modulation Nonlinearities

Amplitude and phase distortion can also result in apparent transmitter overmodulation where none actually exists. Figure 12 is an example of this phenomenon. The antenna system is driven with 80% modulation. In this example, the system has an asymmetric impedance characteristic that boosts the upper sideband while leaving the lower sideband unchanged. The result is a distorted envelope as observed in the far field, with an apparent modulation of 100%. THD is approximately 15%. If the modulation level as observed at the transmitter were increased to 100%, the signal as observed in the far field would have the folded-over waveform of multiplier-type overmodulation and would be greatly distorted in an envelope detector. An engineer with such an antenna system would notice excessive distortion in the field even if the transmitter were modulating normally.

Potentially more damaging to other stations is the case where the modulation level in the far-field is less than that at the transmitter, causing a loudness loss in the field. If the transmitter audio input level were raised in an attempt to increase the far-field modulation, the transmitter would overmodulate. There would be no significant peak modulation increase in the field, although the carrier clipping would cause the average level to increase. The net effect would be a minor increase in average loudness accompanied by a large increase in distortion and possibly adjacent-channel interference.

C. Previous Papers

Most of the effects described above have been recognized for many years and were addressed in depth by Doherty, Moulton, and more recently by Bingeman and Clarke. Doherty, in his classic paper attached as Appendix C, showed how the modulation envelope varies depending on the impedance characteristic at the modulation monitoring point and on whether a voltage or a current sample is taken.[19] He found that modulation percentage measurement required different monitoring point criteria than did sideband power or distortion measurement, and he described how to select these points. He also developed the well-known "line-stretching" technique to rotate the antenna impedance characteristic so that the transmitter output tube sees a symmetrical load.

Moulton, whose paper is attached as Appendix D, published a comprehensive collection of data showing how frequency response, distortion, and square-wave response differed at varying azimuths from a directional antenna. He also showed an example of a folded-over multiplier-type waveform in the minimum of an actual directional antenna.

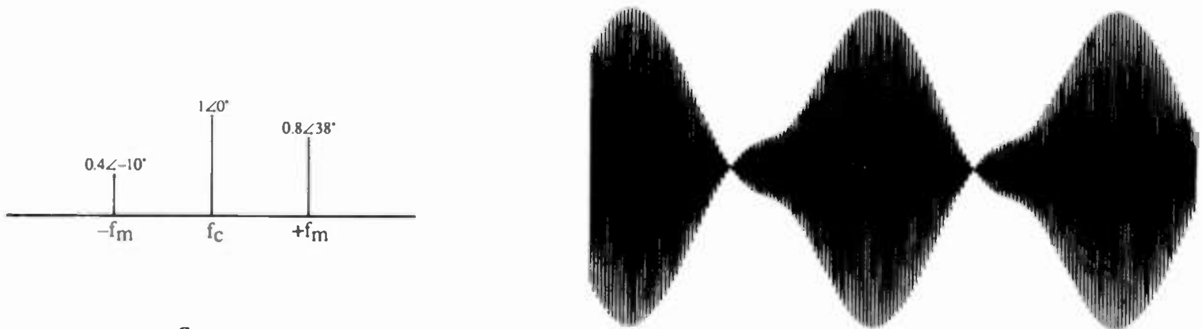
Bingeman and Clarke, in the paper of Appendix E, described a computer technique that quantitatively relates bandwidth to modulation percentage and THD. With this technique, the antenna designer can optimize RF networks to minimize envelope distortion and the difference between transmitter and far-field modulation percentage.

D. Evaluating Station Antenna Performance

A station that experiences distortion on its signal in the field, even though the modulation monitor at the transmitter produces clean audio, or that notices different modulation levels in the field than at the transmitter, is likely to be suffering from antenna system amplitude or phase errors. These errors cannot always be detected from the common point impedance plot of a directional antenna system. Although amplitude and phase errors will be most significant in a narrowband antenna system, they can still occur even in an antenna that has apparently sufficient

[18] The 10 kHz distortion is extremely high because the fundamental component is more than 20 dB below the second harmonic.

[19] The impedance characteristic differs at different points within the antenna phasing and matching circuitry and along the transmission lines.



a. Spectrum

b. Waveform

Figure 12. Example of Apparent Overmodulation.

bandwidth.[20] For example, Bingeman and Clarke (ibid.) describe a relatively broadband antenna system with poor modulation linearity and THD, which was significantly improved by modifying the input matching network.

There are a number of ways for a station to evaluate its performance in this regard. Using an oscilloscope and a modulation monitor, it can examine its modulated envelope in the field at various azimuths while transmitting a cleanly modulated high-frequency tone. Significant discrepancies between the modulation percentages and waveforms seen at the station and those seen in the field are signs of antenna amplitude and phase errors. The station can also compare the outputs of a synchronous detector with an envelope detector at suspect locations. A problem would be indicated if distortion were present in the envelope detector but not in the synchronous detector. The station could also perform the kind of theoretical analysis described in the Bingeman and Clarke paper for various field locations.

IX. Monitoring of Modulation and Occupied Bandwidth

As described in Section VII, extraneous sideband energy can be minimized through proper audio processing system design, but improperly designed or adjusted equipment can cause interference. Unless stations have the ability to properly monitor their modulation characteristics, they have no way of assuring themselves that they are operating as intended. Accurate monitoring equipment and techniques are essential if occupied bandwidth is to be minimized.

A. Transmitter Monitoring

The modulation monitoring point necessary to adjust peak transmitter audio input levels for less than 100% negative modulation must provide an accurate sample of the modulation envelope voltage at the modulated stage of the transmitter, since carrier clipping occurs there. Monitoring the voltage or current waveshape at the wrong point in the transmitter or antenna tuning networks can give an inaccurate indication of the envelope, perhaps showing a lower modulation level than actually exists at the point of modulation. Splatter and distortion would be generated if the modulation level were increased to compensate.

Most transmitters have a modulation monitoring tap for this purpose, but some may not provide accurate voltage envelope samples. A station can evaluate whether its monitoring tap is appropriate by examining the modulation envelope with an oscilloscope while modulating the transmitter with a sine wave. As the input level is increased to 100% modulation and beyond, the carrier should cleanly disappear during negative peaks. If the negative sine wave peaks distort before reaching 100%, or if the negative peaks fold over and then become distorted, an unsuitable monitoring point is indicated. The techniques in Doherty (ibid.) can then be used to select an alternative point that does provide an accurate envelope voltage sample.

B. Need for Field Monitoring Improvements

A station often needs to accurately monitor its modulation and occupied bandwidth at the studio or another field location. It may also need the ability to monitor the signals of other stations. However, it is very difficult to determine a station's modulation percentage in the field or to determine whether the station is in compliance with FCC modulation and occupied-bandwidth limits. Modulation percentage readings on either an oscilloscope or a conventional modulation monitor can be erroneous due to the envelope distortions previously described. While occupied bandwidth can be accurately measured with a conventional swept-filter RF spectrum analyzer if the modulating waveform is noise, such measurements are inaccurate on program material because the filter may miss the transients that are the primary sideband components. If the signal is envelope-detected and analyzed for extraneous audio components, the splatter that might have been transmitted cannot be differentiated from the distortion components that are generated in the envelope detector.

To help solve these monitoring problems, the broadcast industry should develop a high-quality precision demodulator using a synchronous detector. A precision demodulator would allow a broadcast station to accurately and consistently monitor its modulation characteristics throughout its coverage area. The synchronous detector would eliminate envelope distortion, so the received signal would remain undistorted regardless of the monitoring location or the bandwidth characteristics of the antenna system. Any distortion or other spurious components present in the detected audio would be from the station itself.

C. Accurate Occupied Bandwidth Measurements

A synchronous demodulator would make accurate measurements of occupied bandwidth possible, even in a null of a directional antenna. There have been several suggestions concerning such measurements. One was to construct a filter whose shape is as close as possible to the inverse of the FCC occupied bandwidth curve.[21] The output of the demodulator would pass through this filter. Any filter output of greater than the 0 dB reference level

[20] A narrowband antenna is generally apparent from the shape of the plate load impedance curve or the antenna system input impedance curve; examples are a curve which has a complex shape, whose impedance varies greatly across the channel, or which is asymmetric at the plate of the final amplifier.

[21] Personal communication with Robert Orban (id.). For a station with five kilowatts of power or more, this filter would have ideally infinite attenuation below 15 kHz, then would be –55 dB relative to unmodulated carrier from 15 kHz to 30 kHz, –45 dB from 30 kHz to 75 kHz, and 0 dB above 75 kHz. Although such a filter is not physically realizable with such steep slopes, a reasonable approximation could be designed.

would correspond to an emission greater than the FCC limit.

Another suggestion was to analyze the output of the precision demodulator with an audio-frequency FFT analyzer. Such a device uses the FFT techniques of this report on an actual audio signal to digitally compute the spectrum. The audio input is sampled for a predetermined length of time; the computed transform then shows all frequency components of the signal during that time window. With repeated samplings, the maximum occupied bandwidth is accurately determined. This type of analyzer is superior for audio purposes to a swept-filter spectrum analyzer because it captures transients that the filter would probably miss. At present, FFT analyzers are uncommon and quite expensive, but are expected to become more easily available in the future.

A synchronous demodulator would serve other purposes as well. It could be used as a high-quality audio source for the station's monitor system or could be coupled to an audio analyzer for use during proofs-of-performance or other audio tests. If the demodulator were designed with both in-phase and quadrature ("I" and "Q") outputs, the Q output would provide a convenient indication of incidental phase modulation (IPM) in the transmitter. Minimizing IPM is essential for stereo performance, and stations have also found it to be important in improving monaural audio quality.

Accompanying any new occupied bandwidth measurement capability should be a fundamental review of occupied bandwidth measurement procedure, because it is not now well-defined in the FCC Rules.[22] A modulated spectrum is dynamic, with a constantly changing energy distribution. Occupied bandwidth measurements are indications of energy density, so are dependent on whether peak or average energy is measured and on the bandwidth of the measuring device. If more emphasis is to be placed on occupied bandwidth in the future, more rigorous definitions will be required. An appropriate organization such as the National Radio Systems Committee should begin this review.

D. Modulation Percentage Measurement Limitations

To avoid carrier clipping, it would be useful to have an accurate indication in the far field of modulation percentage as observed at the transmitter. Unfortunately, the synchronous demodulator does not provide this capability. Modulation percentage is strictly an envelope parameter, a function of relative envelope amplitudes. Amplitude and phase errors, which are inevitable in the AM transmission process, distort the envelope and make it essentially impossible to determine the transmitted modulation percentage in the field. Stations should recognize that differences between modulation percentage readings in the field and those at the transmitter are perfectly normal. Only the modulation percentage at the proper transmitter monitoring point, described in Section IX-A above, is significant.

X. Conclusions and Recommendations

The material in this report can be used by AM engineers and equipment manufacturers to take actions that will reduce excessive occupied bandwidth in the AM band. By knowing when and how splatter is generated and how to accurately measure modulation conditions, engineers can make appropriate improvements in the design and adjustment of their AM transmitting facilities. These improvements may benefit the station's sound by reducing distortion and listener fatigue. Most importantly, if stations make widespread use of this material to minimize their occupied bandwidth, and if receiver manufacturers become convinced that a high-quality interference-free signal will be increasingly available, it will be in the self-interest of the manufacturers to supply receivers that can accurately reproduce this signal.

The most important conclusions and recommendations of this report are as follows:

1. The primary cause of splatter is excessive high frequency content in modulating audio. This is the single most important conclusion to be drawn from this report. If the modulation contains excessive high frequencies, these will cause splatter. Traditional "overmodulation," meaning the pinchoff of the carrier, is undesirable, but is much less significant than the audio itself. Carrier disappearance, commonly believed to be the significant cause of splatter, has little to do with it.
2. FCC occupied bandwidth limits are bare minimums; they are no guarantee of clean operation. Even egregious modulation practices will normally meet FCC requirements. Stations should strive for occupied bandwidths much narrower than FCC limits. The FCC should consider tightening its limits to improve adjacent-channel interference protection.
3. AM stations can minimize excessive occupied bandwidth through the design of their audio processing systems. All audio processing equipment should contain or be followed by an appropriate overshoot-corrected low-pass filter. A final protective clipper is advised as the last device before the transmitter, preferably as an integral part of the audio processor; this clipper must not be used to increase loudness. To eliminate DC level shift, stations should investigate improving the low-frequency response of their transmitters to approximately 0.1 Hz or less, and transmitter engineers should provide

[22] Section 2.202(a) of the FCC Rules defines occupied bandwidth as the "frequency bandwidth such that, below its lower and above its upper frequency limits, the mean powers radiated are each equal to 0.5 percent of the total mean power radiated by a given emission." However, the Rules are silent on the procedure for measuring occupied bandwidth.

this feature as a field modification and in new models.

4. Amplitude and phase errors in the transmitter and antenna tuning networks can distort the envelope of an AM signal, changing its apparent modulation percentage. To prevent carrier pinchoff, a station must insure that it can accurately monitor the modulated envelope as it exists at the modulated stage of the transmitter. An appropriate modulation monitoring tap is often provided by the transmitter manufacturer, but it may be necessary to choose a different monitoring point using the techniques of the Doherty paper.
5. Envelope distortion due to amplitude and phase errors can affect the quality of the signal received by listeners. Stations should evaluate antenna system modulation performance through field measurements or analysis. Transmitter modulation adjustments should be tailored accordingly; undermodulation might be required at the transmitter to prevent apparent overmodulation in the field. If poor modulation linearity or high envelope distortion is apparent, antenna system improvements should be investigated.
6. A high-quality synchronous detector AM demodulator is needed by the AM industry. Such a demodulator would avoid the envelope distortion caused by amplitude and phase errors and would permit accurate analysis of modulation characteristics in the field. More accurate occupied bandwidth measurements would be possible using such a demodulator. The National Radio Systems Committee or another appropriate organization should begin to develop an improved definition for occupied bandwidth.

For a number of years, AM broadcasters have focused on audio processing and loudness with little concern for the impact of their actions on other stations or on receiver manufacturers. Combined with some basic misconceptions about the causes of splatter, this loudness race has often resulted in signals which sounded good to the stations themselves but wreaked havoc with receivers and other stations. With the current interest in AM improvement, it seems that the entire industry would benefit if each AM station reviewed its operating practices in light of this report.

ACKNOWLEDGEMENTS

The author wishes to thank the following persons for their help in providing equipment and thoughtful comments during the preparation of this report:

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APPENDIX A

DERIVATION OF SYNCHRONOUS DETECTION CHARACTERISTICS

The expanded equation for an amplitude-modulated carrier, modulated with a single tone and passing through a transmission path containing amplitude and phase errors, is

$$e(t) = \cos \omega_c t + A_L \cos [(\omega_c - \omega_m)t + \phi_L] + A_U \cos [(\omega_c + \omega_m)t + \phi_U],$$

where ω_c = carrier frequency,
ω_m = modulating frequency,
A_L = amplitude of lower sideband,
ϕ_L = phase of lower sideband,
A_U = amplitude of upper sideband,
and ϕ_U = phase of upper sideband.

For simplicity we assume a carrier amplitude of 1.

To synchronously detect this signal, it is multiplied by $\cos \omega_c t$.

Because $e(t)$ is a linear summation of three terms, we can multiply each term by $\cos \omega_c t$, using the trigonometric identity

$$\cos A \cos B = 1/2 \cos (A + B) + 1/2 \cos (A - B),$$

and sum the results. For the carrier term,

$$\cos \omega_c t \cos \omega_c t = 1/2 \cos 2\omega_c t + 1/2 \cos 0 .$$

This contains only high frequency and DC terms, which are removed by receiver filtering so can be ignored.

For the lower sideband,

$$\begin{aligned}
&\cos \omega_c t \,(A_L \cos [(\omega_c - \omega_m)t + \phi_L])\\
&= A_L \cos \omega_c t \cos (\omega_c t - \omega_m t + \phi_L)\\
&= 1/2\, A_L [\cos (\omega_c t + \omega_c t - \omega_m t + \phi_L) + \cos (\omega_c t - \omega_c t + \omega_m t - \phi_L)]\\
&= 1/2\, A_L [\cos (2\omega_c t - \omega_m t + \phi_L) + \cos (\omega_m t - \phi_L)].
\end{aligned}$$

Eliminating the left hand high frequency term, the detected lower sideband is

$$1/2\ A_L \cos(\omega_m t - \phi_L).$$

Repeating the same process for the upper sideband yields

$$1/2\ A_U \cos(\omega_m t + \phi_U).$$

Summing the two terms gives the resultant detected signal

$$1/2\ A_L \cos(\omega_m t - \phi_L) + 1/2\ A_U \cos(\omega_m t + \phi_U),$$

which is a pure cosine wave at the modulating frequency, whose amplitude depends on the amplitudes and phases of the individual sideband components.

The detected wave could disappear with certain combinations of phase shifts, but there are no distortion components present. If one of the two sidebands is filtered out at RF frequencies, the phase sensitivity is eliminated, and only amplitude changes will affect the detected output. For a transmission path with no amplitude or phase errors, $A_L = A_U = 1$ and $\phi_L = \phi_U = 0$, so the detected output is

$$1/2 \cos \omega_m t + 1/2 \cos \omega_m t = \cos \omega_m t,$$

which is an exact replica of the modulating signal.

APPENDIX B

FAST FOURIER TRANSFORM PARAMETERS

Throughout the report, FFT examples use a carrier frequency of 625 kHz, with a fundamental modulating frequency of 9765.625 Hz. This corresponds to a carrier period of 0.0016 ms, and a modulating period of 0.1024 ms. These numbers were chosen to represent realistic AM frequencies and to facilitate the computation of direct and inverse Fast Fourier Transforms. With a 1024-point transform, each sample corresponds to 0.0001 ms. The full transform period of 0.1024 ms is one complete cycle of the modulating frequency and 64 complete cycles of the carrier frequency. This choice of frequencies is necessary to prevent undesirable transform artifacts from coloring the output data.

The sideband amplitudes determined by the FFT calculations are valid not only for the frequencies used in the calculations, but for all single modulating frequencies of an AM carrier. For example, if a certain modulation condition resulted in an FFT sideband component of −20 dB at $f_c + 4f_m$ (39,062.5 Hz above the carrier, or 664.0625 kHz), the same modulation condition with carrier and modulating frequencies of 1000 kHz and 1000 Hz, respectively, would also yield a sideband component of -20 dB at $f_c + 4f_m$ (4000 Hz above the carrier, or 1004 kHz).

APPENDICES C, D, E are attached.

Appendix C

Doherty, W.H., "Operation of AM Broadcast Transmitters into Sharply Tuned Antenna Systems," Proceedings of the I.R.E., July 1949, pgs. 729-734.

Operation of AM Broadcast Transmitters into Sharply Tuned Antenna Systems*

W. H. DOHERTY†, FELLOW, IRE

Summary—The impedance of some broadcast antenna arrays varies so much over the transmitted band as to impair the performance of the radio transmitter. The impairment consists in clipping of sidebands and distortion of the envelope at high modulation frequencies. This paper reports on an experimental determination of the nature and magnitude of this impairment and on its substantial reduction by suitable coupling methods.

Introduction

THE IMPEDANCE of a broadcast antenna, and particularly the common-point impedance of an array, often varies widely over the transmission band. When this is the case, the frequency response, amplitude linearity, and modulation capability of the broadcast transmitter can be adversely affected. Recognition of this difficulty by transmitter manufacturers has led to the formulation of an RMA specification for the "normal load" into which a transmitter should operate and meet its performance requirements. This is a load whose resistance does not depart more than 5 per cent from its midband or carrier-frequency value at ±5 kc or 10 per cent at ±10 kc, and whose reactance, which is zero at midband, does not exceed 18 per cent of the midband resistance at ±5 kc or 35 per cent at ±10 kc.

A study of the effect of frequency-sensitive loads from the viewpoint of the transmitter designer has been carried out by engineers of the broadcast transmitter development group of Bell Telephone Laboratories under the supervision of J. B. Bishop. Extensive data and oscillograms have been taken which indicate the extent of the impairment of transmitter performance under a variety of conditions, and the effectiveness of corrective methods which are to be described.

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I. Monitoring Methods

The first phase of the study necessarily involved determination of proper monitoring conditions whereby actual sideband power, effective percentage modulation, and true distortion in the signal delivered to the antenna system can be measured. A preliminary discussion of the monitoring problem appeared in a previous publication,[1] in which it was shown that the appearance of the modulation envelope is greatly different at different points in a coupling circuit or at different points along a transmission line when the impedance of the termination at the side frequencies differs substantially from the impedance at the carrier frequency. This is illustrated by the oscillograms of Fig. 1, which show the voltage envelope for a modulation frequency of 7500 cps as observed at three points along a transmission line or coupling network whose termination, for example, is equivalent to a series-tuned circuit resonant at the carrier frequency. At the termination, or at points removed therefrom by an even number of quarter wavelengths, a fully modulated voltage wave may appear (Fig. 1(a)), and the amplitude of each of the two side-frequency voltages accordingly will be one-half the amplitude of the carrier voltage. But, because the impedance rises on either side of the carrier frequency (Fig. 1(d)), the side-

[1] W. H. Doherty, "Notes on modulation of AM transmitters," *The Oscillator*, p. 22, no. 5; October, 1946.

frequency *currents* will each be less than one-half the carrier current, and an inspection of the current envelope would show substantially less than 100 per cent modulation. On the other hand, at points removed from the termination by odd quarter wavelengths, the impedance will correspond to that of a parallel-tuned circuit (Fig. 1(f)), decreasing on either side of the carrier frequency; and, although the current envelope if observed would show a fully modulated wave, the voltage

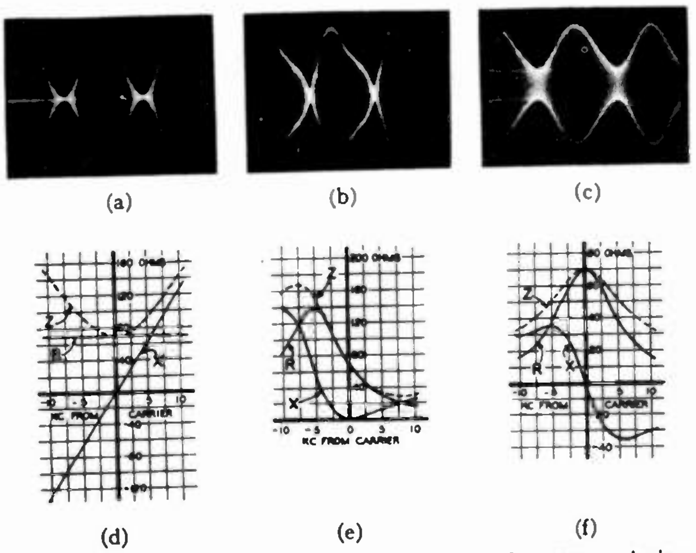


Fig. 1—Voltage envelopes and impedance versus frequency relations at three points in an output circuit or transmission line with narrow-band termination.

envelope (Fig. 1(c)) shows considerably less than 100 per cent modulation—in the case illustrated, only 60 per cent, since the impedance at the side frequencies is only 60 per cent of the impedance at the carrier frequency (the inverse of the situation of Fig. 1(d)). Finally, at odd eighth-wavelength points, where the impedance versus frequency curve is dissymmetrical (Fig. 1(e)), the voltage envelope has the distorted appearance indicated in Fig. 1(b). The current envelope at this point would also be badly distorted. A monitoring rectifier and conventional distortion-measuring instrument would register a high percentage of distortion for this wave, yet there are no extraneous side frequencies being radiated, the envelope distortion being entirely due to the inequality and phase dissymmetry of the two desired side-frequency voltages at this particular point in the line or coupling circuit. It is obvious that if, in addition, the operator were to raise the audio input level, endeavoring to bring about apparent full modulation at this point, much more severe distortion would be registered, because the wave at other points would then be over-modulated.

The impedance versus frequency relations indicated are those corresponding to a simple resonant circuit in which the ratio of reactive volt-amperes to watts at 550 kc, for example, is approximately fifty to one. The impedance curve of a simple resonant circuit appears on a Smith chart[2] as a circle. Fig. 2 shows impedance circles D, E, and F for points along a transmission line

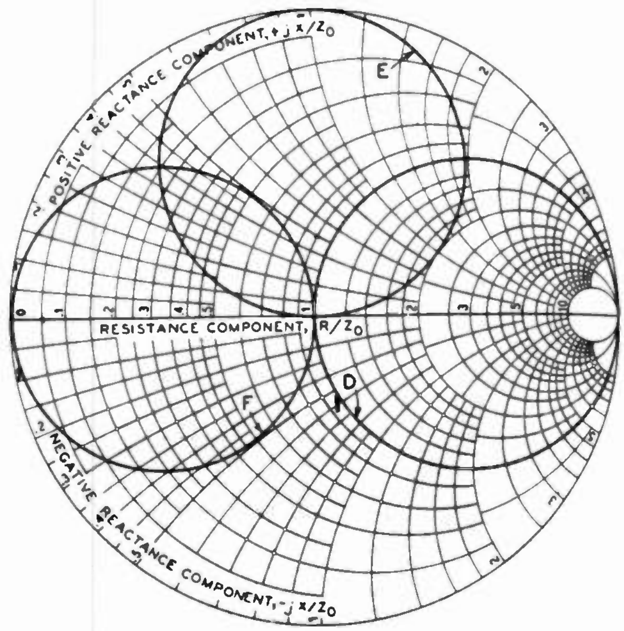


Fig. 2—Smith-chart impedance diagrams for a simple resonant circuit at points along a transmission line corresponding to D, E, and F, of Fig. 1.

or coupling circuit where the impedance versus frequency relations would correspond to D, E, and F of Fig. 1. The frequency dependence in the case shown is several times more severe than the RMA standard for artificial antennas for transmitter testing, but is comparable with that frequently found in actual broadcast installations.

Measurement of Effective Modulation and Distortion

If one wishes to determine actual delivered sideband power by measurement of a sample of the current envelope, it is necessary to make the inspection and measurement at a point in the circuit corresponding to Fig. 1(d), where the series resistance is independent of frequency over the band transmitted, since it is the current squared times the series resistance that determines power. We may refer to such points as D points for convenience. In order that the measurement may include true radiated distortion power, i.e., power in extraneous sidebands, the constancy should hold over a correspondingly wider band. Now, when measuring at such points, one should not try to adjust the audio input for 100 per cent modulation of the current envelope observed on the oscilloscope, but for a certain lower percentage—60 per cent, in the case considered in Fig. 1—since the voltage envelope (not being observed by the operator, but shown in Fig. 1(a)) will then have reached full modulation, and any further increase

[2] P. H. Smith, "An improved transmission line calculator," *Electronics*, vol. 17, pp. 130–134; January, 1944.

would necessarily entail severe distortion in both the voltage wave and the current wave.

If, on the other hand, it is the voltage envelope rather than the current envelope that is to be monitored, the point of measurement should be one where the *parallel* rather than the series resistance is constant over the transmitted band, since the power is the voltage squared divided by the parallel resistance. It can be shown[3] that this constancy will be found only at points corresponding to Fig. 1(f), one-quarter wavelength removed from D points. We may label these as F points. At these F points one must not look for a fully modulated wave, since the *current* wave (not being observed) will have reached full modulation, in the case considered, when the voltage wave, seen in Fig. 1(c), is only 60 per cent modulated.

Thus we have the curious situation that, with narrow-band antennas, and when modulating at high audio frequencies, only certain points D in the output circuit of the transmitter are suitable for determination of effective sideband power and distortion power when a sample of the current wave is being analyzed, and only certain other points F when a voltage sample is being analyzed; while the maximum permissible modulation of the wave being analyzed has to be set in the reverse manner, i.e., by inspection of the voltage envelope at points D or the current envelope at points F, or by calculation from the impedance versus frequency curve.

It should be noted that at D points, where the series resistance is independent of frequency, the parallel resistance is greater for the sidebands than for the carrier frequency, and a distortion measurement on the voltage envelope would be pessimistic since a given distortion power will be represented by a disproportionately high sideband voltage. Similarly, at F points, where the parallel resistance is independent of frequency but the series resistance is lower for the sidebands than for the carrier, a distortion measurement on the current envelope will be pessimistic. Thus the point where minimum distortion is registered is the correct monitoring point for the ideal case described, and will also, in general, afford the most reliable measure of distortion power in cases where the impedance diagram is irregular.

In order to permit the plotting of conventional curves of distortion versus modulation frequency for particular percentages of modulation, it is necessary to express the percentage modulation in terms of the quantity—current or voltage—which can be allowed to attain full modulation at the monitoring point, even though the true sideband power, with sharply tuned loads, does not correspond to full modulation. There is, moreover, justification for this in the fact that, when the maximum permissible modulation (without overmodulation) is reached, the power amplifier tubes are being required to deliver either full peak current or full peak voltage to the load, even if not both.

While it is only in certain types of programs that a broadcast transmitter is subjected to heavy modulation at high audio frequencies, the application of a test tone and measurement of harmonic distortion at such frequencies is a part of regular testing routine carried out with standard station equipment, and, when correctly done, provides valuable information both in the initial tune-up of a transmitter and in maintenance. However, an equally important application of the monitoring technique just described is in connection with the over-all frequency characteristic of the transmitter from audio input to sideband power output. This will be discussed in Section II. With the recent establishment by the Federal Communications Commission of a requirement for submission of performance data prior to renewal of licenses, it is important to be able to verify delivery of appropriate energy to the array from the transmitter for all modulation frequencies.

II. Operation of the Power Amplifier

With the impedance-versus-frequency characteristic varying widely from point to point in a coupling circuit, it is scarcely necessary to state that the characteristic as found at the particular point where the power-amplifier tubes are connected is of profound importance in the performance of the amplifier, regardless of the type of circuit or modulation method used. In particular, reverting to Fig. 1, if the impedance seen by the tubes were to vary with frequency in the manner given by Fig. 1(e), the tubes would *have* to impress on the circuit a voltage envelope resembling Fig. 1(b) in order that the voltage envelope at a proper voltage monitoring point (an F point) might be free of distortion. Indeed, since the impedance dissymmetry of Fig. 1(e) necessarily entails considerable phase modulation in the radio-frequency wave which does not show up in the oscillogram, it would be necessary for the tubes to introduce a corresponding phase modulation as well as to deliver a voltage envelope distorted in amplitude. Since one can ask only that a transmitter deliver to its load a voltage wave or a current wave free of phase modulation and having its envelope identical in shape to the audio input wave, it is apparent that, when the load is frequency-sensitive, only a point of impedance symmetry such as a D point or an F point is appropriate for making connection to the tubes. In the former case, when a high-frequency test tone is applied, the tubes will be asked to deliver full rated voltage and less than full rated current at the peak of the envelope; in the latter case, full rated current at less than full rated voltage. In either case, undistorted voltage and current envelopes are desired.

[3] The well-known impedance-inverting property of quarter-wave lines and their equivalent networks can be expressed in a form which shows, interestingly, that if one speaks of parallel components at one end and series components at the other end, the inversion holds for the resistances and also, *independently*, for the reactances; hence, the constant series resistance at D points necessarily means constant parallel resistance at points one-quarter wavelength therefrom, irrespective of reactance values.

Tests on several transmitters at powers from 1 to 10 kw, with very sharply tuned artificial antennas, have confirmed that when arrangements are made to connect the tubes to a D or F point the distortion at high modulation frequencies, when properly measured, differs very little from the distortion measured with a flat antenna, the slight increase observed being attributable to the effect of the sharp antenna on the width of the band over which negative feedback is effective. Fig. 3 shows the harmonic distortion in an experimental 10-kw broadcast transmitter at 95 per cent modulation with (1) a flat dummy antenna, and (2) a dummy

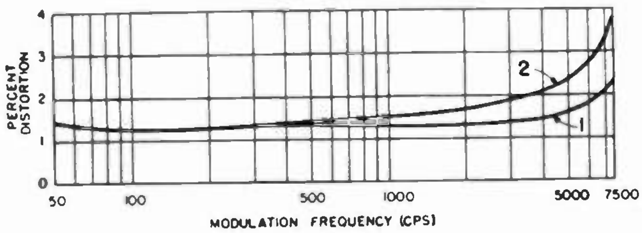


Fig. 3—Distortion curves for an experimental 10-kw transmitter at 95 per cent modulation with flat antenna (curve 1) and frequency-sensitive antenna (curve 2).

antenna having a ratio of kva to kw of 25 to 1 at 550 kc, giving an impedance-versus-frequency characteristic about half as severe as that of Fig. 1. The tubes were connected at an F point and the distortion and per cent modulation were determined in the manner described. In contrast, with the tubes connected at a point of impedance dissymmetry and the distortion and per cent modulation improperly monitored, apparent distortions as high as 20 and 30 per cent were recorded. Fig. 4 gives an example of the kind of envelope shape that was observed under such conditions. A distortion-measuring instrument registered 21 per cent for this wave.

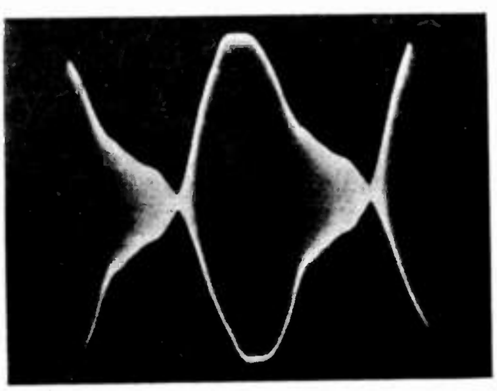


Fig. 4—Modulation envelope of a 10-kw transmitter for frequency-sensitive load as seen under improper operating and monitoring conditions.

The transmitter used for this test employed the high-efficiency circuit[4] devised by the author, with grid-bias modulation[5] of the final stage and employing "envelope" feedback, in which a sample of the radio-frequency output is detected and fed back to the audio input circuits.

From a distortion standpoint, the D and F connections are found to be about equally satisfactory. For transmitters employing envelope feedback, the F connection offers an important advantage in that it provides automatic compensation for the sideband-clipping tendency of the antenna. This comes about from the fact that with envelope feedback it is desirable for reasons of bandwidth to "pick off" a radio-frequency voltage sample directly at the plate of the final power tube; and if this point is an F point, the sample will then represent the voltage across a parallel resistance that is the same for the sidebands as for the carrier, and consequently the feedback will act to maintain a flat frequency characteristic in the radiated sideband power. When the impedance-versus-frequency characteristic of an antenna differs from that of a simple tuned circuit (i.e., exhibits in the transmitted band a curvature differing from that of the circles in Fig. 2), the corrective action of the feedback is less complete but still substantial. To achieve equivalent compensation for a narrow-band antenna characteristic by the use of high-kva coupling meshes in the output circuits would be unduly expensive in apparatus and would involve critical tuning and considerable radio-frequency power loss.

The potency of feedback derived from the radio-frequency envelope in bringing about delivery of the desired sideband energy to the antenna system is shown in Fig. 5, which pertains to the same 10-kw transmitter and the same types of loads to which the distortion curves of Fig. 3 apply. Curve 1 of Fig. 5 gives the overall frequency characteristic of the transmitter at 50 per cent modulation with no feedback when operating into a broad-band resistance load. The departures from flatness at the low and high ends arise mainly in the audio circuits in the transmitter. Curve 2, still without feed-

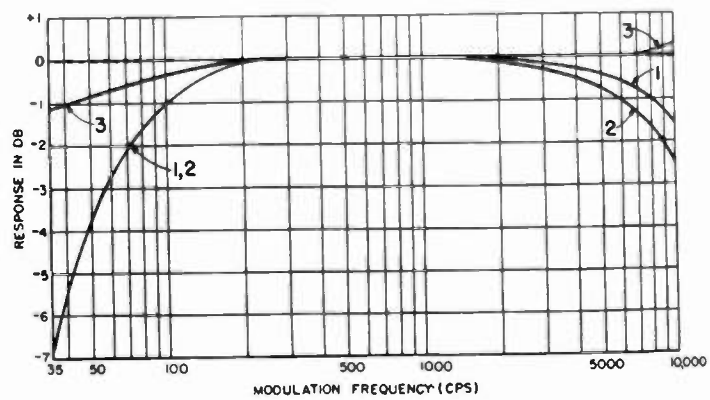


Fig. 5—Frequency characteristic curves for a 10-kw transmitter.

back, includes the additional loss at high modulation frequencies due to the frequency-sensitive load when the tubes are connected at an F point, the monitor being also connected at this point. But with envelope feedback applied, as derived from the voltage at this F point where the parallel resistance is constant over the band, the improved performance indicated by curve 3 is ob-

[4] U. S. Patent No. 2,210,028, W. H. Doherty.
[5] U. S. Patent No. 2,226,258, H. A. Reise and A. A. Skene.

tained. Fig. 6 shows, in contrast, corresponding curves for connection of the power amplifier tubes at a D point, where the parallel resistance is higher for the sidebands than for the carrier (but with the monitor still connected at an F point, since it is only here that a voltage-operated monitor will give a true indication of sideband power). The feedback in this case, being derived from

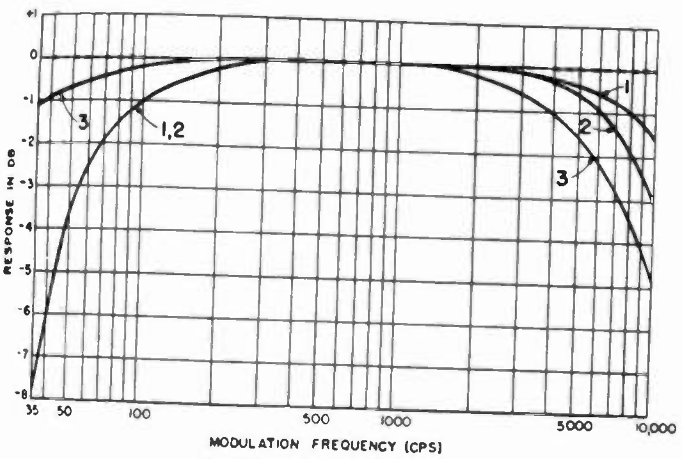


Fig. 6—Effect of unfavorable feedback connection on the frequency characteristic.

the voltage at the D point where the tubes are connected, actually aggravates the sideband-clipping action of the sharp antenna, as seen in curve 3 of Fig. 6. This is because, while the power tubes tend to impress higher than normal sideband voltages on the circuit on account of higher impedance to sideband frequencies, and thus partially compensate for the clipping action of the circuit, the feedback acts to prevent this compensation.

III. Transmitter Output-Circuit Design

To incorporate in broadcast transmitters the facilities for reorientation of the impedance-versus-frequency characteristic of any sharply tuned load that may be encountered, a variable phase shifter is required, equivalent to a "line stretcher," with a total range approaching 180 degrees to cover all cases. Such a phase shifter, as built for the experimental 10-kw transmitter on which these tests were conducted, is shown in Fig. 7(a). With the recent commercial availability of variable vacuum capacitors of wide capacitance range, linear calibration, and high voltage rating, it was most practical to incorporate this phasing device in the high-impedance output circuit of the transmitter prior to transforming down to the transmission line. The transformation ratio of the phase shifter shown is unity, and the coils L_1 and L_2 have reactances equal to the terminating resistance R regardless of the phase shift desired. For the minimum phase shift of 90 degrees, capacitor C_2 likewise has a reactance of R ohms, and C_1 and C_3 have zero capacitance. For greater phase shifts, capacitor C_2 is increased, and capacitors C_1 and C_3 come into play. By proper choice of these three capacitances, the phase shift can be increased to any value up to 270 degrees (or more) with no change required in the inductances, the input impedance remaining a pure resistance of R ohms through-

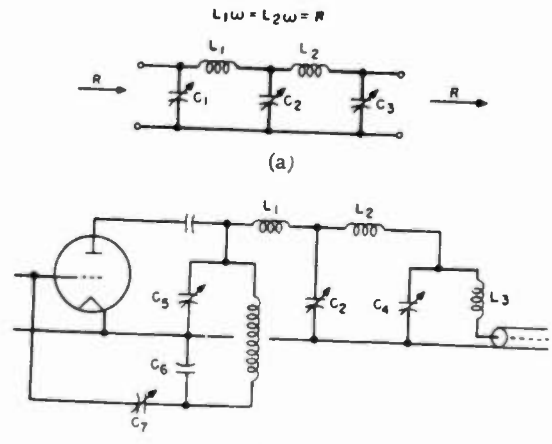


Fig. 7—Incorporation of a phase shifter in the transmitter output circuit.

out the range. The required admittances for these capacitances are:

$$C_1\omega = C_3\omega = \frac{1}{R}\left(1 - \cot \tfrac{1}{2}\Phi\right) \tag{1}$$

$$C_2\omega = \frac{2}{R}\left(1 - \tfrac{1}{2}\sin \Phi\right) \tag{2}$$

where Φ is the phase shift desired. These relations are shown in Fig. 8.

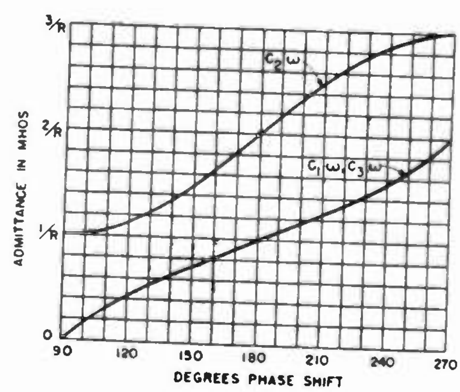


Fig. 8—Capacitive admittance required for the phase shifter of Fig. 7.

This phasing device is inserted, as shown in Fig. 7(b), between the radio-frequency plate terminal of the power amplifier and a load circuit consisting of C_4 and L_3, normally tuned to resonance and matching the amplifier to the transmission line. The built-out neutralizing circuit shown contains a "tank" capacitor C_6 with which the power tube is tuned in the conventional manner. With this arrangement, when a phase shift other than 90 degrees is required for improving performance with a frequency-sensitive antenna, the capacitances C_1 and C_3 of Fig. 7(a) are obtained by simply increasing the values of C_4 and C_6, which, like C_2, are variable vacuum capacitors.

In stations whose radiation patterns are different for day and night operation, the daytime pattern usually involves a simpler excitation of the antenna array, giving a flatter impedance characteristic. The phase-shifter adjustment chosen would, accordingly, be that best fitted to the nighttime impedance curve.

In the 10-kw transmitter built for testing these principles, the tube shown in Fig. 7(b) was the No. 2 or "peak" tube of a high-efficiency amplifier operating at a plate potential of 10,000 volts. The desired load impedance for the amplifier was 720 ohms. Coils L_1 and L_2 were accordingly made 720 ohms each and coil L_3 was adjusted for 186 ohms to obtain, in combination with C_4 a transformation from the 51.5-ohm coaxial-line impedance to 720 ohms. Capacitor C_2 is made 720 ohms in all cases where the antenna presents no bandwidth problem. When a narrow-band antenna is encountered, the required phase shift for best operation is determined by plotting the impedance characteristic at the input terminals of the transmission line on a Smith chart with an added peripheral scale, as shown in Fig. 9. Recalling

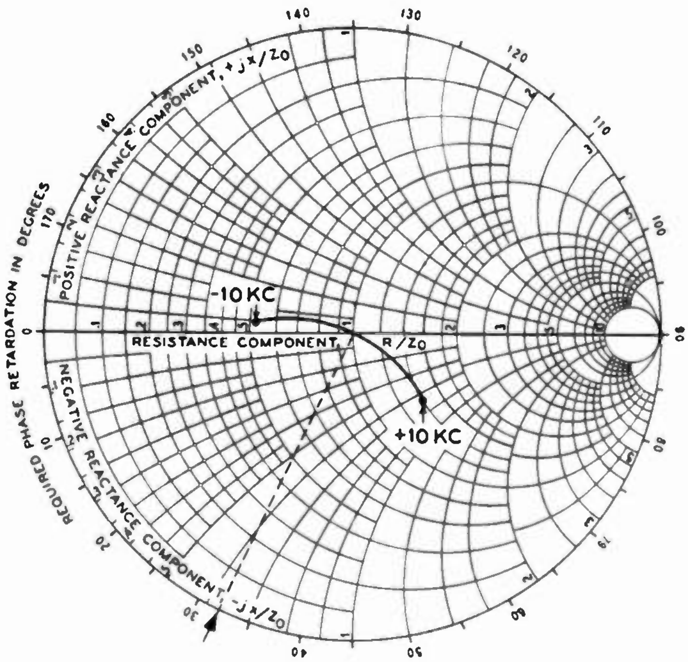


Fig. 9—Determination of the required phase shift in a typical case from the Smith chart.

that the desired orientation of the characteristic at the plate of the power-amplifier tube is that of circle F of Fig. 2, it is seen from Fig. 9 that the total phase retardation desired between the transmission line and the power tube for the case illustrated is either 32 degrees, or 180 plus 32 degrees. Since the phase retardation introduced by coil L_3 is $\tan^{-1} 186/51.5$ or 75 degrees, it is necessary to adjust the phase shifter to 137 degrees to obtain the total of 212 degrees required orientation. Fig. 8 then gives the values of C_2 and for the increments to be made in C_4 and C_6 to constitute effectively the capacitances C_1 and C_3 of Fig. 7(a). The actual final adjustment of C_6 is, of course, that which gives a unity power-factor load at the amplifier tube.

The radio-frequency plate terminal, being an F point, is used as the source of energy for the feedback rectifier and monitoring rectifier, due consideration being given to the fact that with a narrow-band antenna the voltage envelope observed when modulation is applied at high audio frequencies will not indicate 100 per cent modulation when the current delivered by the tubes is fully modulated.

With some antenna arrays the impedance may vary so irregularly with frequency as to call for a compromise adjustment which is difficult of prediction from the impedance diagram. In such cases, experimental determination of optimum phase shift is desirable by direct observation of the envelope shape at the plates of the tubes. The type of phase shifter described is especially well adapted to this procedure because of the wide range of adjustment possible without removal of power and the constancy of carrier-frequency impedance as the phase shift is varied.

Because of the extra harmonic suppression provided by the phase-shifting network, the usual harmonic filter connected at the input to the transmission line and employing mica capacitors (due to the low impedance) is no longer required. In cases where an unusually high degree of suppression is needed for one harmonic, a small fixed vacuum capacitor paralleling L_3 will provide a substantial increase in suppression. The circuit described thus combines with its property of handling frequency-sensitive loads the features of high harmonic suppression and long-life components.

In most cases, a station with a new antenna and transmitter would operate initially with the phase shifter set for its minimum shift of 90 degrees. After the completion of all antenna adjustments and the establishment of the final radiation pattern, the transmission-line input impedance would be measured over a wide band, and the desirability of phase correction determined. By merely increasing the capacitances of three variable capacitors, any additional phase shift desired can then be introduced to permit optimum performance of the transmitter and provide a monitoring point where the most reliable measurements of this performance can be made.

Acknowledgment

The experimental transmitters used in the investigations reported were built by H. A. Reise and C. W. Norwood of Bell Telephone Laboratories, who were also responsible for all of the performance data obtained. The contributions of these engineers and their colleagues are gratefully acknowledged.

Appendix D

Moulton, Clifford H., "Signal Distortion by Directional Broadcast Antennas," Proceedings of the I.R.E., May 1952, pgs. 595-600.

Signal Distortion by Directional Broadcast Antennas*

CLIFFORD H. MOULTON†, ASSOCIATE, IRE

Summary—**Directional broadcast antenna systems are inherently capable of producing signal distortion. One type of distortion, described by Doherty,[1] involves the frequency sensitivity of the antenna input impedance and relationships between this impedance and the transmitter and transmission lines. A second type of distortion results from the directional radiation characteristics of the array, and is caused by differences in the response conditions of a directional array for the carrier and each sideband frequency.**

Sources of Signal Distortion

THE WAVEFORM of the modulation envelope is responsible for the receiver audio-signal waveform in a double sideband AM system. The modulation envelope must therefore remain unchanged during transmission if signal distortion is to be avoided. Any change in the relative phase or amplitude of a signal component may result in a change in the waveform of the modulation envelope, which may cause audio distortion.

The directional pattern of an array is a function of the transmitted frequency, and is necessarily different at the sideband frequencies from that at the carrier frequency. These radiation-pattern differences cause changes in the amplitudes and relative phases of the signal components arriving at the receiving point and hence changes in the modulation envelope. The effect of altering the phases or amplitudes of the signal components in any particular manner may be determined by adding the components vectorially and obtaining the distorted modulation envelope.

Certain combinations of phase shifts and amplitudes result in large amounts of modulation-envelope distortion. One such condition occurs in directions where the high-frequency sideband amplitudes are increased with respect to the carrier amplitude. When the transmitter is modulated 100 per cent with a high-frequency tone, the sideband amplitudes are then greater than required for 100-per cent modulation of the carrier, producing a type of overmodulation.

At low audio modulating frequencies the response conditions for the carrier and sideband frequency components are essentially identical. With high audio modulating frequencies and relatively low carrier frequencies, however, the antenna bandwidth may be sufficiently low to result in severe changes in the relative phases and amplitudes of the signal components. In antenna-pattern null directions the strong carrier-frequency radiation fields of the antennas may almost completely cancel, but the cancellation may rapidly approach reinforcement for a sideband component as its frequency is removed from the carrier frequency. Carrier-frequency null directions are therefore most likely to be accompanied by increased high-frequency sideband power. In directions of maximum carrier power the converse conditions are likely to occur, resulting in reduced high-frequency sideband power. The audio-frequency response characteristics of a directional array will therefore be a function of the receiving direction.

It is significant that although audio distortion components may be found in the receiver output when modulation-envelope distortion exists the array itself does not introduce new frequency components in the transmitted signal but merely alters the amplitudes or phases of existing components. The receiver second detector is responsible for the addition of the large number of distortion components found in the receiver output.

Experimental Measurements

The simplest method of evaluating the audio distortion and frequency-response changes produced by directional broadcast arrays appears to be by direct measurement rather than by calculations or by the use of scale models. Two standard broadcast stations with directional antenna systems were therefore selected as test stations. One of these, station A, employed a two-tower 550-kc array with shunt feed. The other, station B, employed a three-tower 1,280-kc array with series feed.

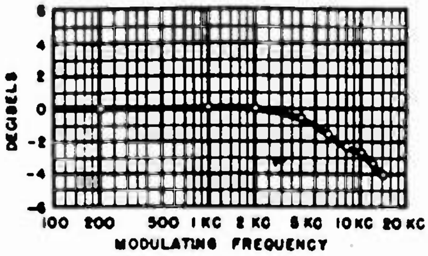
The signal distortion and frequency-response data which follow were obtained during the test period from 1 to 6 A.M. A battery amplifier with a whip antenna and ground rod was carried into completely open spaces. The received test signal was amplified at carrier frequency by broad-band amplifiers, and sent to a mobile unit by coaxial cable. After further amplification at the truck, the signal was then distributed to a Tektronix model 511-AD oscilloscope for photographing waveforms and to a Hewlett Packard model 330-B noise and distortion meter for distortion- and frequency-response measurements. Receiving locations were chosen outside of the antenna induction-field regions and at random distances from the antenna systems.

Frequency-response data are for 50-per cent modulation at the transmitter. This modulation percentage allows a few decibels of sideband power increase before the point of severe distortion, and yet prevents noise and interference from becoming too objectionable. Two-

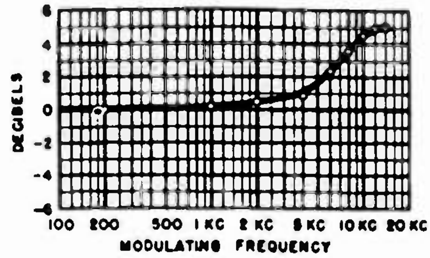
* Decimal classification: R326.4×R148.11. Original manuscript received by the Institute, January 22, 1951; revised manuscript received, October 1, 1951.

† Tektronix, Inc., P. O. Box 831, Portland 7, Ore.

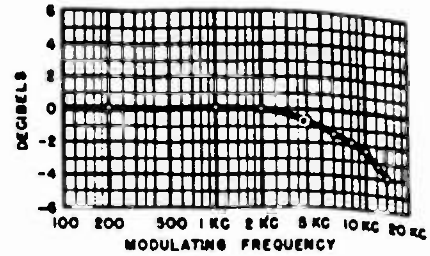
[1] W. H. Doherty, "Operation of AM broadcast transmitters into sharply tuned antenna systems," Proc. I.R.E., vol. 37, p. 729; July, 1949.



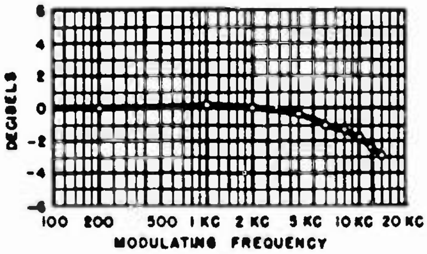
30° azimuth angle
Carrier amplitude 135 per cent of rms.



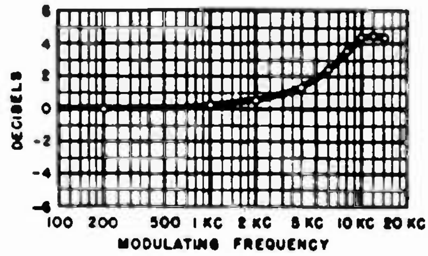
150° azimuth angle
Carrier amplitude 24 per cent of rms.



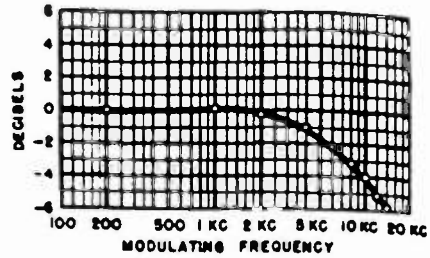
270° azimuth angle
Carrier amplitude 139 per cent of rms.



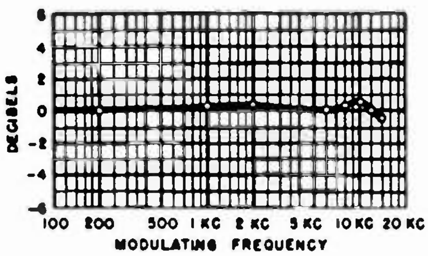
60° azimuth angle
Carrier amplitude 129 per cent of rms.



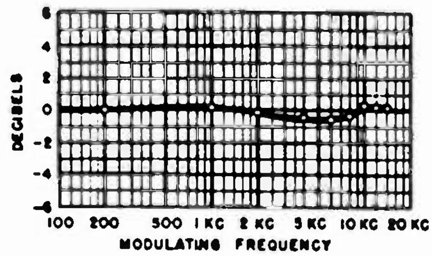
180° azimuth angle
Carrier amplitude 25 per cent of rms.



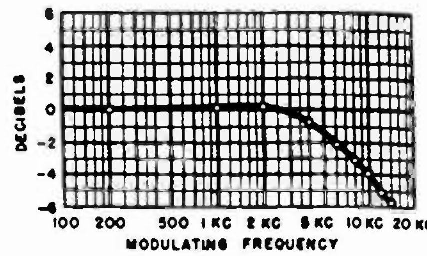
300° azimuth angle
Carrier amplitude 120 per cent of rms.



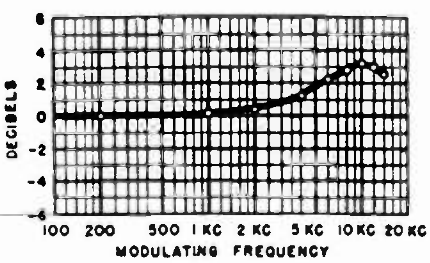
90° azimuth angle
Carrier amplitude 83 per cent of rms.



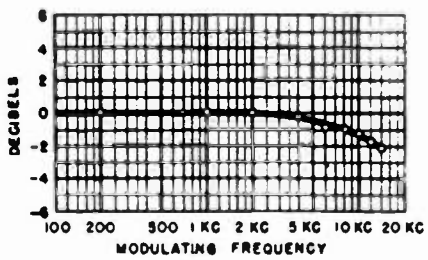
210° azimuth angle
Carrier amplitude 63 per cent of rms.



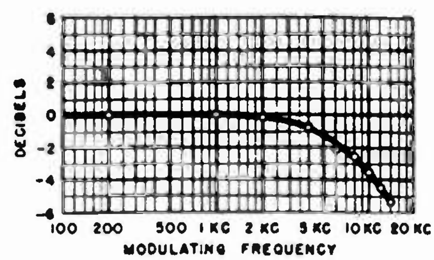
330° azimuth angle
Carrier amplitude 102 per cent of rms.



120° azimuth angle
Carrier amplitude 33 per cent of rms.

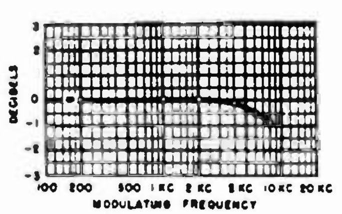


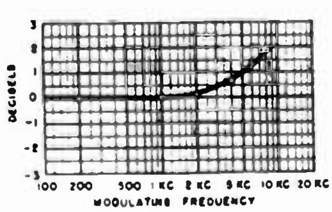
240° azimuth angle
Carrier amplitude 117 per cent of rms.

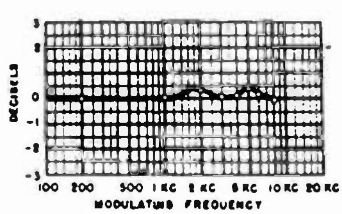


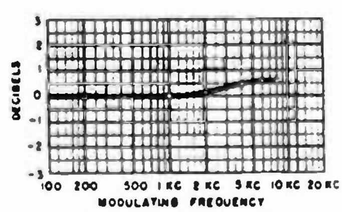
360° azimuth angle
Carrier amplitude 112 per cent of rms.

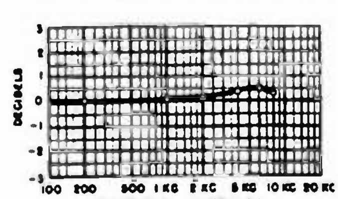
Fig. 1—Audio-frequency response curves for station A at various azimuth angles. Data are for 50-per cent modulation at the transmitter and random distances from the antenna system.

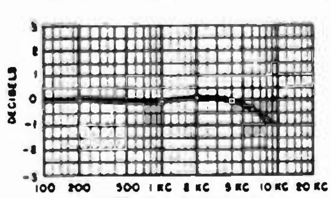

25° azimuth angle
Carrier amplitude 6.6% of rms.

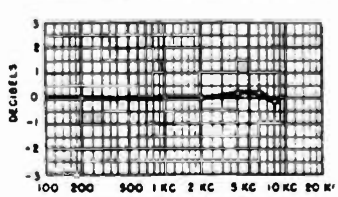

35° azimuth angle
Carrier amplitude 3.3% of rms.


48° azimuth angle
Carrier amplitude 5.5% of rms.


90° azimuth angle
Carrier amplitude 26% of rms.


270° azimuth angle
Carrier amplitude 161% of rms.


327° azimuth angle
Carrier amplitude 132% of rms.


125° azimuth angle
Carrier amplitude 12% of rms.

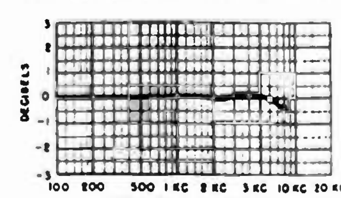

167° azimuth angle
Carrier amplitude 21% of rms.

Fig. 2—Audio-frequency response curves for station B at various azimuth angles. Data are for 50-per cent modulation at the transmitter and random distances from the antenna system.

kc, square-wave audio modulation was employed for a series of waveform photographs at various receiving locations because the phases and amplitudes of a large number of audio-frequency components could be observed simultaneously. Test data at station B were limited to 8 kc because of modulator overload.

Results

Audio-frequency response curves at various azimuth angles are shown for station A and station B in Figs. 1 and 2, respectively. The high-frequency audio response is greater in the null directions than in the maximum directions in both cases. The zero decibel level for each direction is taken as the received audio level at 200 cycles for 50-per cent modulation at the transmitter output. The differences in the response curves at station B are much less than at station A, partly because station B operates at a carrier frequency 2.33 times as high as station A. The audio spectrum represents a much smaller percentage of the carrier frequency at 1280 kc than at 550 kc. It is necessary to correct the data of station B to 550-kc conditions, in order to compare them directly with the data of station A, by dividing the modulating frequencies of station B by 2.33. Tests at station B extend to 8 kc, which becomes 3.44 kc when corrected.

The maximum spread in response curves, Δ, expressed in decibels, is plotted as a function of audio modulating frequency for both stations in Fig. 3. Another curve of corrected Δ for station B is given in Fig. 3 after correcting the audio frequencies to 550 kc. From the latter curve it is seen that Δ would be higher for station B than

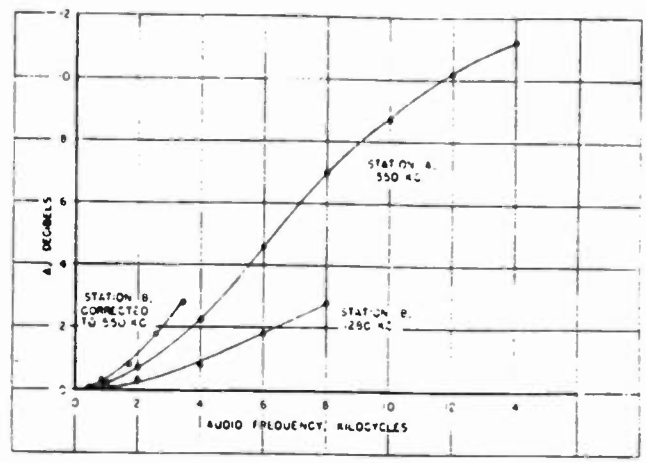
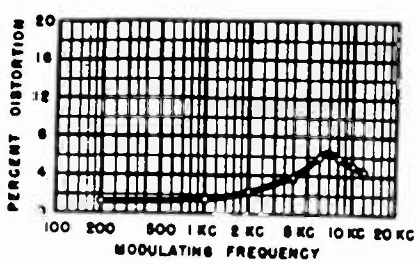
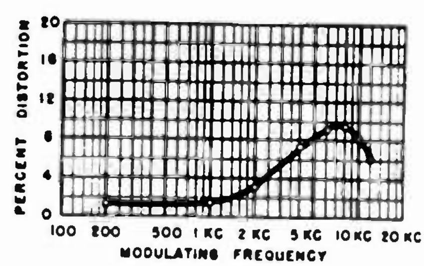


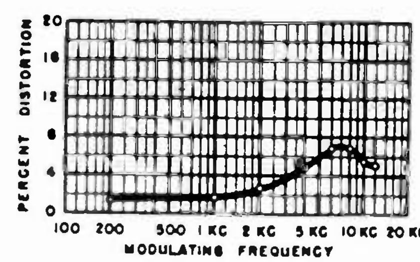
Fig. 3—Maximum response-curve differences, Δ, versus modulating frequency for 50-per cent modulation at the test stations.



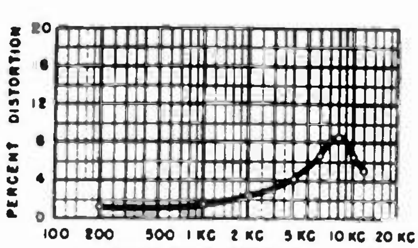
30° azimuth angle
Carrier amplitude 135 per cent of rms.



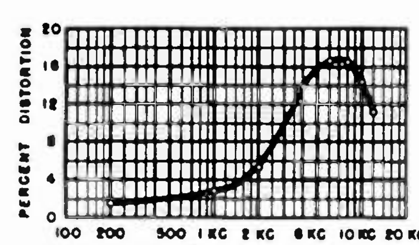
135° azimuth angle
Carrier amplitude 24 per cent of rms.



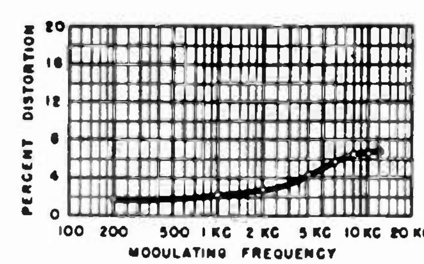
255° azimuth angle
Carrier amplitude 124 per cent of rms.



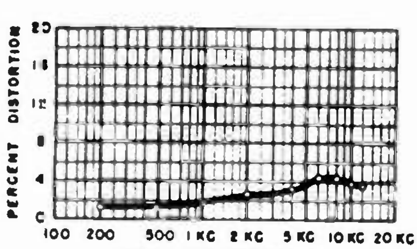
75° azimuth angle
Carrier amplitude 110 per cent of rms.



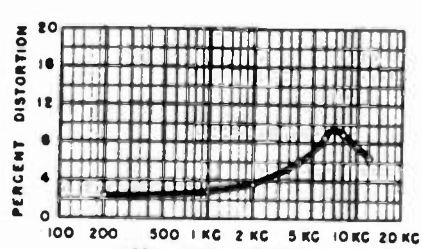
165° azimuth angle
Carrier amplitude 24 per cent of rms.



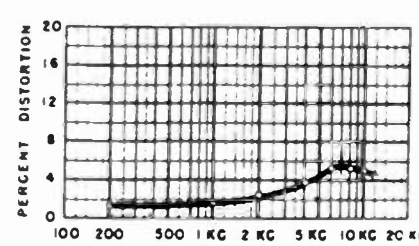
300° azimuth angle
Carrier amplitude 120 per cent of rms.



120° azimuth angle
Carrier amplitude 33 per cent of rms.



210° azimuth angle
Carrier amplitude 63 per cent of rms.



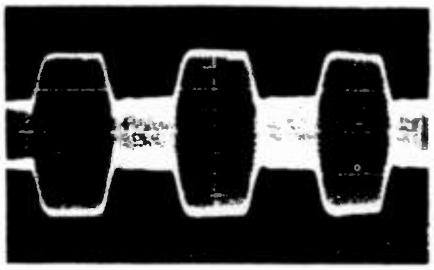
345° azimuth angle
Carrier amplitude 104 per cent of rms.

Fig. 4—Station A audio-frequency harmonic distortion as a function of modulating frequency at various azimuth angles. The transmitter was modulated 50 per cent.

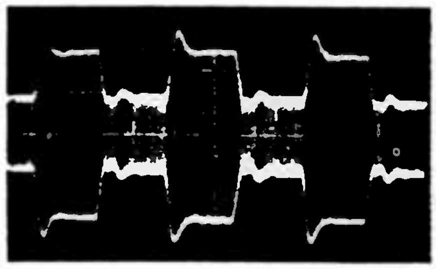
station A if both were to be scaled to operate at 550 kc.

Harmonic distortion of the received signal with 50-per cent sinusoidal modulation at the transmitter output is shown in Fig. 4 as a function of modulating frequency and azimuth angle at station A. The highest distortion percentages are found in the null directions. Similar data could not be taken at station B because of interference and noise in the null directions.

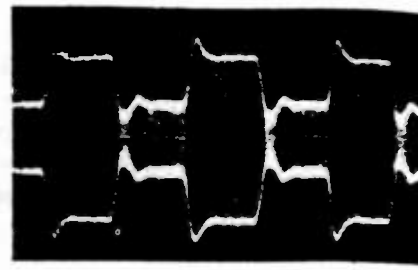
The responses of both stations to square-wave audio modulation are seen in Figs. 5 and 6. The waveforms of station A are markedly different in the null and maximum directions. The slow rises of the waveforms in the maximum directions show reduced high-frequency sideband power while the leading- and trailing-edge overshoots in the null directions show increased high-frequency sidebands. The peculiar waveform in the leading



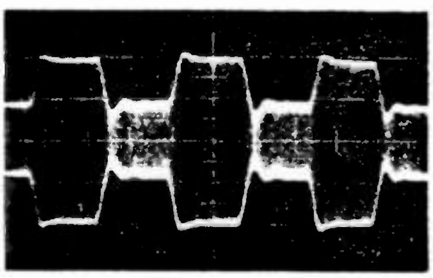
45° azimuth angle
Carrier amplitude 139 per cent of rms.



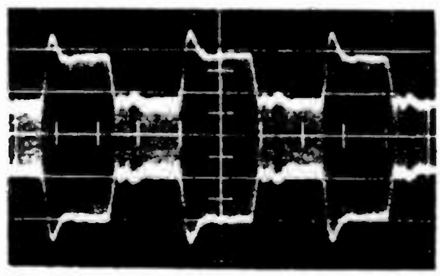
135° azimuth angle
Carrier amplitude 24 per cent of rms.



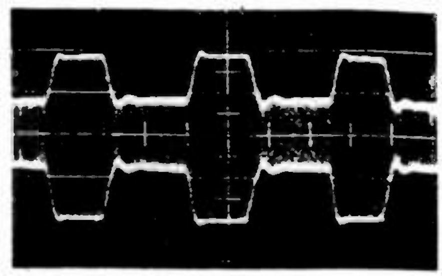
180° azimuth angle
Carrier amplitude 25 per cent of rms.



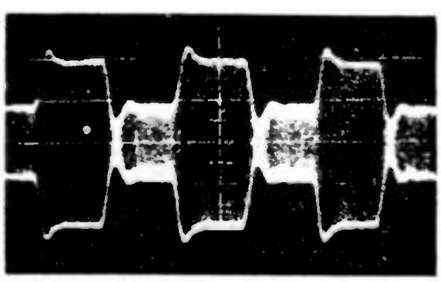
90° azimuth angle
Carrier amplitude 83 per cent of rms.



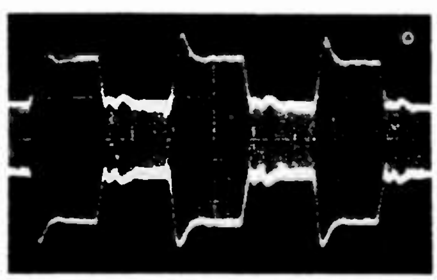
150° azimuth angle
Carrier amplitude 24 per cent of rms.



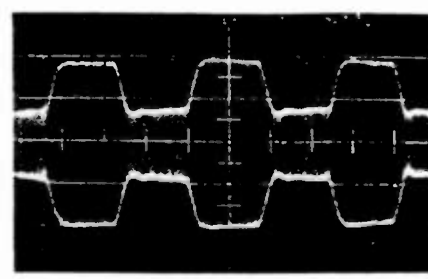
210° azimuth angle
Carrier amplitude 63 per cent of rms.



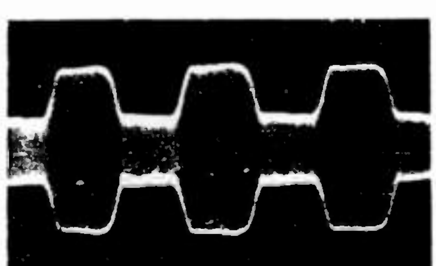
120° azimuth angle
Carrier amplitude 33 per cent of rms.



165° azimuth angle
Carrier amplitude 24 per cent of rms.

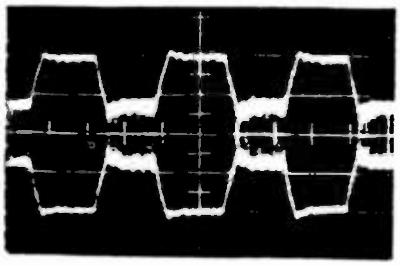


270° azimuth angle
Carrier amplitude 139 per cent of rms

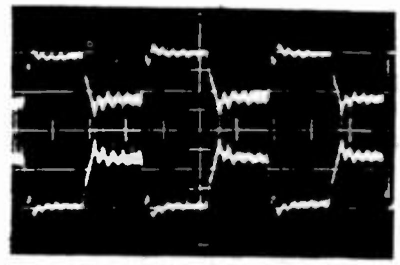


345° azimuth angle
Carrier amplitude 104 per cent of rms.

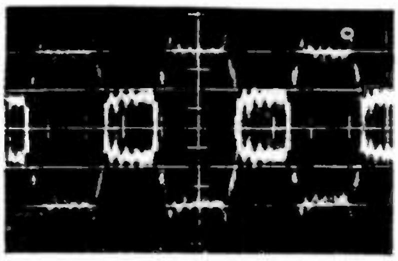
Fig. 5—Photographs of station A modulation-envelope oscilloscope traces at various azimuth angles for 2-kc square-wave modulation. The transmitter was modulated 50 per cent.



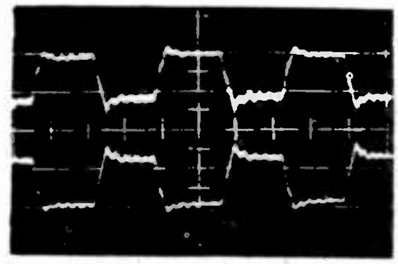
25° azimuth angle
Carrier amplitude 6.6 per cent of rms.



48° azimuth angle
Carrier amplitude 5.5 per cent of rms.

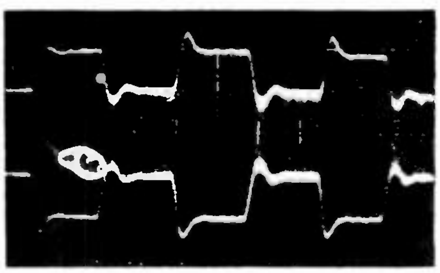


35° azimuth angle
Carrier amplitude 3.3 per cent of rms.

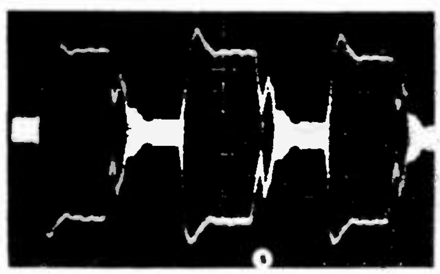


270° azimuth angle
Carrier amplitude 161 per cent of rms.

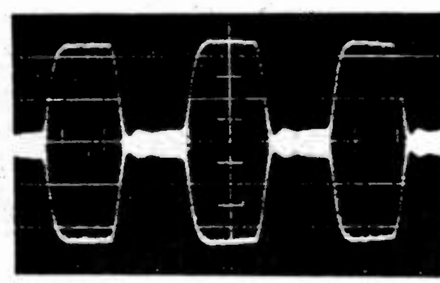
Fig. 6—Station B square-wave modulation photographs for various azimuth angles. The transmitter was modulated 50 per cent with a 2-kc square wave.



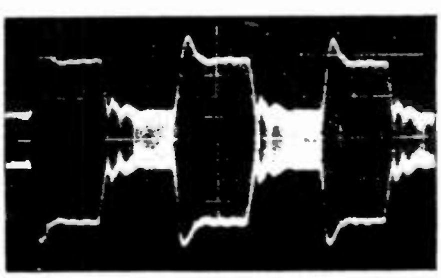
(a)
Modulation 40 per cent.



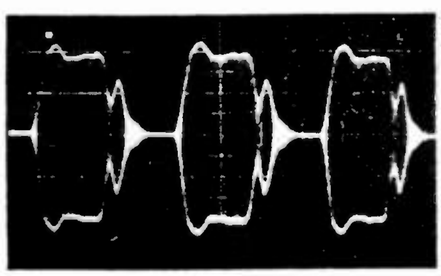
(c)
Modulation 85 per cent.



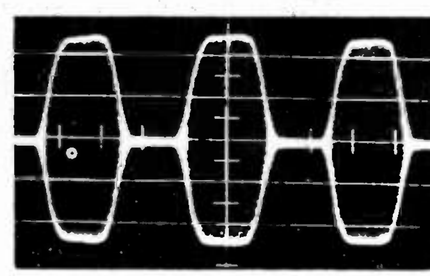
(e)
Modulation 85 per cent.



(b)
Modulation 60 per cent.



(d)
Modulation 98 per cent.



(f)
Modulation 95 per cent.

Fig. 7(a–d)—Station A square-wave response for various modulation percentages at an azimuth angle of 135° (carrier amplitude 23.8 per cent of rms).
Fig. 7(e and f)—Same at an azimuth angle of 255° (carrier amplitude 133.5 per cent of rms).

edge of the downward modulation half of the square wave in the null region of station A should be noted. The damped oscillation on each waveform of station B in Fig. 6 was present with a dummy antenna and was produced in the transmitter audio system. Noise and interference are responsible for the fuzziness of the null direction waveforms.

Fig. 7 shows the effect of modulation percentage on the transient response characteristics of station A for two receiving directions. A given percentage of modula-

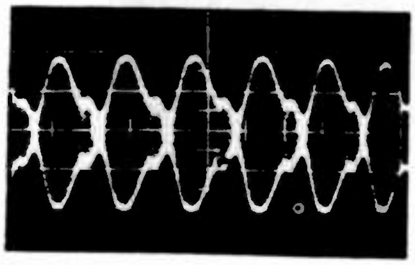
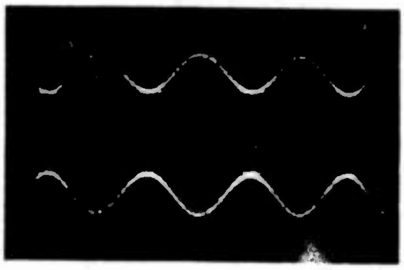
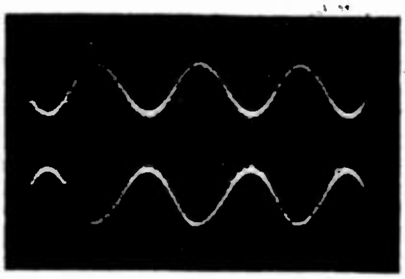


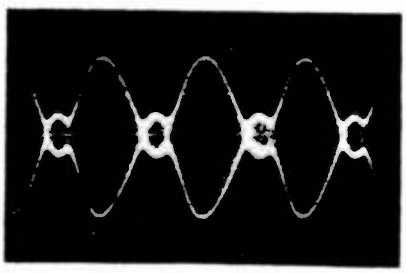
Fig. 8—Modulation envelope waveform at station A for 70-per cent modulation at 4 kc at an azimuth angle of 165° (carrier amplitude 23.8 per cent of rms).



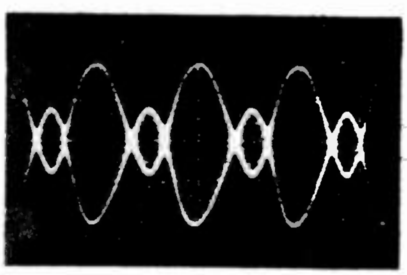
Transmitter modulation 20 per cent.



Transmitter modulation 40 per cent.



Transmitter modulation 60 per cent.



Transmitter modulation 80 per cent.

Fig. 9—Station A modulation-envelope photographs for 10-kc sinusoidal modulation at various modulation percentages. The receiving location azimuth angle was 135° (carrier amplitude 23.8 per cent of rms).

tion results in entirely different modulation envelopes in the two receiving directions. High percentages of modulation are accompanied by severe distortion in the null region.

An example of serious distortion in the null direction at 4 kc is shown in Fig. 8. Analysis of this waveform shows that the two sidebands are not of the same amplitude and have been shifted slightly in time phase with respect to the carrier. Waveforms of the type shown in Fig. 9 occur when the sidebands have similar amplitudes and are shifted from the normal phase position. The effect of modulation percentage on the modulation envelope of station A in the null direction with 10-kc sinusoidal modulation is seen from Fig. 9.

Discussion

Arrays with many elements, nigh-Q tuning networks, negative power elements, and deep nulls are more likely to have severe antenna distortion than simpler, lower Q systems. Stations operating at the low-frequency end of the broadcast band are much more subject to this distortion than those operating at the high-frequency end of the band.

Deep nulls should be avoided from a standpoint of signal distortion if service is to be rendered in the null directions.

Directional signal distortion in the horizontal plane would not be expected for single-element vertical antennas.

While this article has dealt principally with transmitting antenna signal distortion at broadcast frequencies, it is apparent that somewhat similar effects should also be expected for receiving antennas, for other radio frequencies, and for other types of modulation.

Conclusions

1. Signal distortion is observed in directional broadcast antennas, and is found to be a function of receiving direction.
2. Signal distortion results from changes produced by the directional antenna system in the magnitudes or relative phases of the signal components.
3. Directional signal distortion is accentuated by deep nulls, low-percentage antenna bandwidth, high audio-modulating frequency, and high percentage of modulation.

Acknowledgment

The author wishes to thank the co-operating radio stations for the use of their facilities, Professor Grant S. Feikert of Oregon State College for his guidance, and the many students and staff members of the Oregon State College Physics and Electrical Engineering Departments who assisted in this project.

Appendix E

Bingeman, Grant W. and C.V. Clarke, "AM Antenna System Bandwidth versus Harmonic Distortion," IEEE Transactions on Broadcasting, June 1977, pgs. 50-55.

AM ANTENNA SYSTEM BANDWIDTH VERSUS HARMONIC DISTORTION

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With the advent of AM stereo, the growing concern for high-fidelity, and the race for loudness, bandwidth has become a popular topic.[1] We all know that the state-of-the-art transmitter has solved the transmitter bandwidth problem, and that the weakest link in the chain is the antenna system, now more than ever. This paper will show that most AM antenna systems can be designed and adjusted to have a bandwidth compatible with today's transmitters.

There are two critical monitoring points in the AM RF chain that are affected by antenna system bandwidth: the final stage of the transmitter, and the far-field detector. Since the need for an optimized bandwidth at the transmitter has been recognized for several decades,[2] this paper emphasizes antenna system bandwidth as it relates to distortion at the far-field detector.

This analysis includes several assumptions. These are:

1. The receiving point is located in the major lobe of the pattern, and in the far-field;
2. The detector responds to the peak of the envelope only;
3. Antenna radiation efficiency is constant over the band of interest;
4. Antenna current distribution is sinusoidal;
5. The transmitter is ideal and is a separate, distortionless voltage generator for each frequency

   $(f_o, f_o - f_m, f_o + f_m)$;
6. The transmission lines are lossless;
7. The transmitter is being modulated by a 10 kHz tone;
8. The center frequency (carrier) is 750 kHz.

In order to meaningfully analyze the data, a computer program was developed by C. V. Clarke (Senior Engineer, Harris Broadcast Products Division), which quantitatively relates bandwidth to total harmonic distortion (THD) and modulation index. This program uses a numerical Fourier series approach[3] to calculate the amplitude of the first five harmonics produced at the detector due to an imperfect sideband pair.

The approach to data collection was to model several variations of a representative phasor on the computer (the results of this analysis also apply to nondirectional systems). The complex sideband currents in each radiator were determined, and then combined at the receiving point (P, fig. 1). From this superposition the r.m.s. THD and the degrees of positive and negative modulation at the far-field detector were determined for a given modulation index at the transmitter. The idea was to minimize the difference between the modulation index at the transmitter and the modulation index in the far-field, and to minimize the harmonic distortion in the far-field. The modulation index and THD seen at the transmitter will rarely be the same as those seen in the far-field.

The following parameters (and all the others they imply) were modeled as being frequency dependent:

1. All the reactances;
2. Radiator self impedances;
3. Radiator mutual impedance;
4. Transmission line electrical length;
5. Far-field angular distance, ϕ.

The radiators were modeled as a tee network (fig. 2) in order to eliminate the problem caused by interdependence of radiator currents and base impedance (eqn. 1 and 2).

$$Z_1 = Z_{11} + Z_{12}I_2/I_1 \qquad \text{eqn. 1}$$

$$Z_2 = Z_{22} + Z_{21}I_1/I_2 \qquad \text{eqn. 2}$$

For the phasor, a typical two-tower (fig. 3) array producing a cardioid pattern (fig. 4), using steel-guyed, uniform cross-section, series-fed towers of 90 degrees electrical height, was chosen.[4] The spacing and field ratios and angles were chosen so as to produce good final base impedances, within the constraints of the pattern. The tower self impedances were taken from empirical data. It was assumed that any impedance-transforming antenna accessories (e.g.: static drain chokes) or stray reactances within the system had already been taken into account, that all components were lossless, and that there was no mutual coupling between coils (even in the power divider).

An input tee network to match the transmitter to the power divider was chosen, as opposed to an input 'L' network, in order to allow optimization of the sideband response seen by the transmitter. It should be noted, however, that the Smith chart model of this 'line stretcher' is not exact if all the sideband impedance points are lumped together on the same chart. This is because the lower sideband wavelength we are dealing with is significantly longer than the upper sideband wavelength, but the angle scale around the circumference of the Smith chart is valid at one frequency only. More importantly, a tee network model of a transmission line is valid at only one frequency. Therefore, computer iteration was used to obtain exact and conclusive results.

It was determined that minimum far-field distortion is obtained when the sideband impedance, as seen by the transmitter, approaches symmetry; that is, when the sideband reactances are equal and opposite, and when the sideband resistances are equal.

This was not a particularly surprising result, and tends to support the validity of the analysis up to this point.[5] As a matter of fact, it looks like there is no reason why a well-designed phasor cannot be adjusted to introduce much less harmonic distortion (for a given, single tone) than the rest of the system building-blocks.

But we have yet to deal with far-field modulation index (defined in fig. 5).

The following are the original sideband impedances at the input to the phasor, after the system has been adjusted to produce the desired pattern:

$$Z\ (f_0 - 10\ \text{kHz}) = 57.1 + j13.4$$

$$Z\ (f_0) = 50.0 + j0$$

$$Z\ (f_0 + 10\ \text{kHz}) = 36.2 + j2.77$$

This is a fairly typical sideband pair. The phase shift across the input matching network is -90 degrees.

Figures 6 and 7 show what happens to the envelopes when we try for 100% negative modulation at the transmitter. The far-field envelope exists along the locus of points equidistant from the two towers. In the absence of any non-linearities, distortion occurs because of the unequal propagation characteristics across the frequency band of interest. Since the transmitter is an ideal voltage generator, a current pick-up was used for the envelope in Figure 6. A voltage pick-up would not indicate any distortion at this point in the circuit, except during negative overmodulation.

Figure 8 displays the harmonic distortion at the two monitoring points as a function of modulation index seen at the transmitter point. It is considerably worse for the listening audience in the far-field, and this is by no means the worst listening point in the pattern for the present unsymmetrical sideband situation.[6]

Figure 9 compares the modulation indexes at the transmitter with those in the far-field. There is almost a one-to-one relationship between the positive indexes, but the negative index relationship is markedly non-linear. The reference line is for the case of a perfectly flat, resistive load across the band of interest.

After iterating a while by altering the phase shift across the input matching network, a relatively symmetrical, realizable sideband pair at the transmitter for a phase shift of -155 degrees was obtained:

$$Z\ (f_0 - 10\ \text{kHz}) = 58.6 - j25.0$$

$$Z\ (f_0) = 50.0 + j0$$

$$Z\ (f_0 + 10\ \text{kHz}) = 56.6 + j20.1$$

Incidentally, when the phase shift across the input tee is changed like this, the radiator sideband voltages and currents change, but the radiator sideband impedances remain fixed, since we are dealing with a linear system.

With 100% modulation at the transmitter (both positive and negative), the THD becomes a mere 0.5%. The corresponding far-field distortion is only 0.1%. The reason for the smaller far-field THD is the fact that the far-field modulation index is only 66%, for a transmitter index of 100%. The envelopes are shown in Figures 10 and 11.

Figure 12 displays the modulation index relationship for the improved sideband pair. As you can see, the relationship is linear, but the far-field loudness seems to have dropped. Distortion is way down, but so is the far-field modulation index. This could get someone in a bit of trouble when the next rating period rolls around, but fortunately there is a solution.

There is another point of sideband symmetry available when the input matching network is adjusted to have a phase shift near -55 degrees. When this is done, the sideband impedances are as follows:

$$Z\ (f_0 - 10\ \text{kHz}) = 41.9 + j9.43$$

$$Z\ (f_0) = 50.0 + j.0$$

$$Z\ (f_0 + 10\ \text{kHz}) = 40.8 - j11.2$$

Figure 13 displays the new modulation index comparison; still linear, but now the far-field index is greater than its transmitter counterpart. Figure 14 shows the transmitter envelope when 100% modulation exists in the far-field. This new situation requires that the transmitter modulation index be limited to less than 85%, in order not to overmodulate in the far-field with a 10 kHz tone. One might conclude from all this that a real-world transmitter can now operate in a region where its inherent distortion is lower, its power consumption is lower, its tube life is extended, and component reliability is raised. Real transmitters, however, are designed to work best into a flat resistive load. As antenna system bandwidth is increased, the impedances of the balanced sideband pair will approach the center frequency impedance.

As the frequency of the modulating tone is decreased, line A in Figure 13 will move closer to the reference line, in most cases. For example, a 5 kHz tone will require a modulation index greater than 85% to produce 100% modulation at the far-field point. Since normal programming energy is concentrated closer to the center frequency than 10 kHz, the deleterious effects of bandwidth will not be as pronounced with average program material. But, of course, the FCC proof of performance is not done with average program material.

If we move the far-field monitoring point a full 90 degrees of azimuth to point P_2 (fig. 4), we find that the modulation index is about five percent higher, and the THD is about the same (compared with point P). Similar results are obtained for a 45 degree change in azimuth, at point P_3 (fig. 4). It is fortunate that this array produces only a five percent variation across the major listening area; if the variation were large, there would be little to gain from optimization of the sideband pair back at the transmitter.

As for the rest of the phasor, we can decrease the 'Q' ($Q = X/R$) of the common-point center frequency impedance by increasing the power divider inductance. Some people say that this will help the system bandwidth. But first, let's look at the sideband impedances at the input to the power divider, for our second design (input tee -155 degrees):

$$Z\ (f_0 - 10\ \text{kHz}) = 22.7 + j4.28$$

$$Z\ (f_0) = 26.9 + j9.12$$

$$Z\ (f_0 + 10\ \text{kHz}) = 36.2 + j6.62$$

This is about what would appear at the input to a 3/8 wavelength transmission line terminated in a series resonant circuit.

But when we increase the power divider inductance in an attempt to reduce 'Q', we also increase the impedance looking into, and the lagging phase shift across, the power divider. Therefore, we must change the phase shift of the phase-adjusting network (in this case, resulting in a reduction in component reactances), and adjust the input matching network (an increase in component reactances). These necessary additional changes may actually negate any improvement obtained at the power divider, so it is important to treat each phasor individually.

We must keep in mind that a bandwidth 'bottleneck' may exist elsewhere in the phasor. If we treat anything other than this bottleneck first, there may not be an appreciable improvement in the overall system bandwidth.

There are several potential bottlenecks in a phasor:

1. The input matching network,
2. Power divider,
3. Phase-adjusting network(s),
4. Transmission lines,
5. Antenna coupling networks,
6. And the radiators.

The networks (1,2,3,5) can be optimized by keeping the phase shifts across them as low as practical (there are trade-offs with stability[7] and ease of adjustment), and by using the least number of components possible, including zero. A transmission line can be modeled as a series of tee networks (one for each frequency), and thus fits in with Item 3. The radiator bandwidth is a bit more complicated, being a function of effective tower diameter, electrical height, spacing, system base currents, the ground system, and nearby structures. In most cases, the radiators are the bottleneck.

The approach to optimization of AM antenna system bandwidth is to first select the radiator configuration that provides the most cost-effective bandwidth for a given pattern requirement, using the tee-network model (fig. 2). Once this is done, it is a simple matter to optimize the bandwidth of the rest of the system, if one is methodical in developing the specific computer model, which should also include the exact output network of the transmitter.

In conclusion, we have found that the far-field harmonic distortion is minimized when the THD at the final stage of the transmitter is minimized (when sideband symmetry is approached), and that it is not necessary to get especially close to symmetry to obtain excellent distortion figures. More importantly, we have found that an improvement in far-field distortion may be accompanied by a considerable decrease in the far-field modulation index at all levels of modulation (for a single tone). We have also been reminded that a single Smith chart is not an exact model for the behavior of the sideband pair in a lumped-parameter system, and that an iteration technique can overcome this problem. It has been pointed out that sideband symmetry is not a panacea for real transmitters; thus an attempt to improve the system bandwidth should also be made. Finally, we have seen that it is possible to have our cake and eat it, too: vanishing distortion, perhaps a decreased workload for the transmitter, and loudness can all be obtained by following the methods outlined in this paper.

FOOTNOTES

1. Since this paper deals with harmonic distortion only, the reader interested in phase and intermodulation distortion is referred to:

   Stanford Goldman, Frequency Analysis, Modulation and Noise, Chapter entitled "Modulation", 1948, McGraw-Hill.
2. W. H. Doherty, "Operation of AM Broadcast Transmitters Into Sharply Tuned Antenna Systems", July, 1949, Proceedings of the I.R.E.
3. Skilling's Electrical Engineering Circuits, 1952, John Wiley and Sons, NY, Pages 439 - 455.
4. Preliminary design for a 50 kW phasor for Dr. Ivo Facca, Alvorada, Brazil.
5. Bill McCarren, Associate Director of AM Transmission Systems, CBS Radio, Paper presented at 1976 NAB Show in Chicago, "Antenna Q vs. Audio Response".
6. Clifford H. Moulton, "Signal Distortion by Directional Broadcast Antennas", May, 1952, Proceedings of the I.R.E.
7. R. S. Bush, "Phasing System Network Sensitivities", Available from Harris Broadcast Products Division, Quincy, Illinois.

FIG. 1. FAR-FIELD CURRENT SUPERPOSITION

$$I_P = I_1 e^{j\phi_1} + I_2 e^{j\phi_2}$$

$I_i = I_i e^{j\theta}$ = tower$_i$ base current

ϕ_i = angular distance

lines ϕ_1 and ϕ_2 are parallel, for P in the far-field

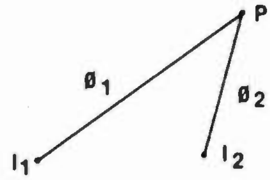


FIG. 4. RADIATION PATTERN

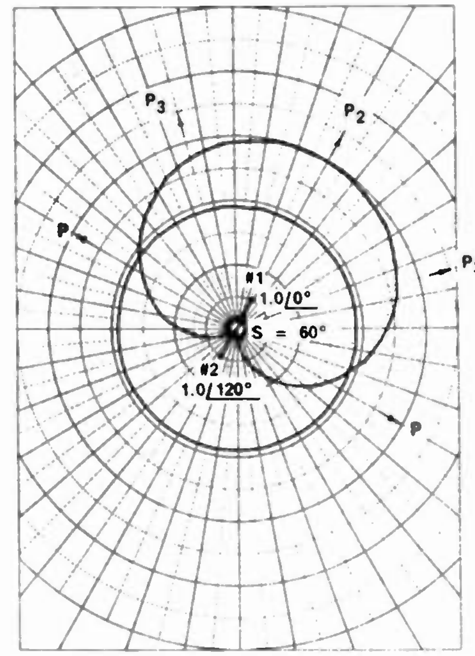
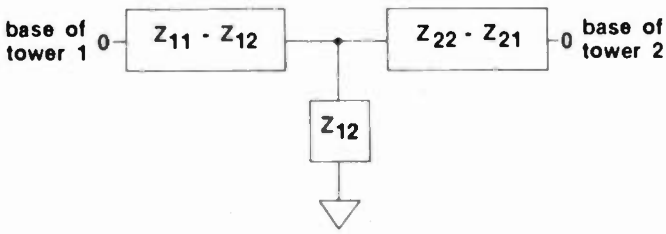


FIG. 2. MODEL OF RADIATORS



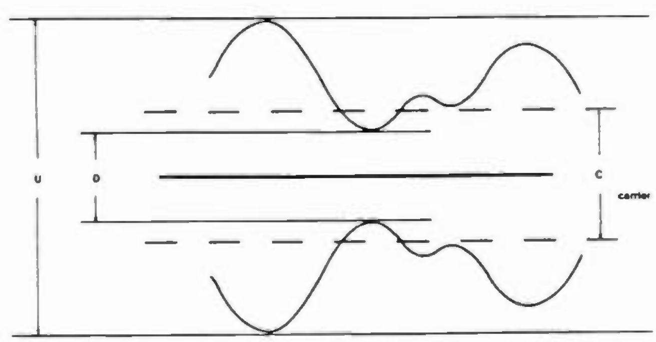
Z_{11} = self-impedance of radiator 1

Z_{22} = self-impedance of radiator 2

$Z_{12} = Z_{21}$ = mutual impedance between radiators 1 and 2

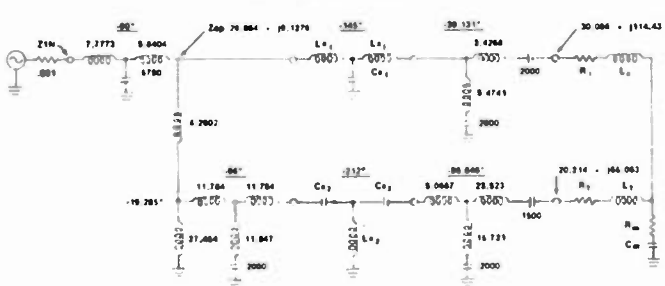
For this analysis, $Z_{11} = Z_{22}$.

FIG. 5. MODULATION INDEX DEFINED



positive modulation = (U - C)/C
negative modulation = (C - D)/C

FIG. 3. PHASOR R.F. DRAWING



$L_1 = L_2$

$R_1 = R_2$

$R_1 + jWL_1 = Z_{11} - Z_{12}$

$R_2 + jWL_2 = Z_{22} - Z_{21}$

COIL UNITS — uH

CAP UNITS — pF

FIG. 6. ENVELOPE AT TRANSMITTER - DESIGN #1

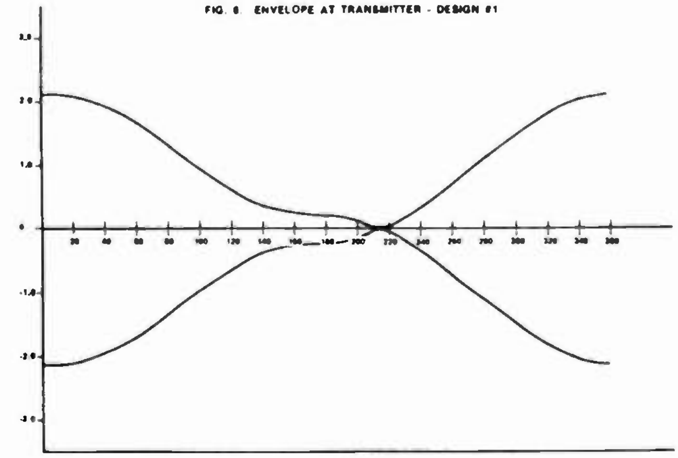


FIG. 7 ENVELOPE AT FAR FIELD - DESIGN #1

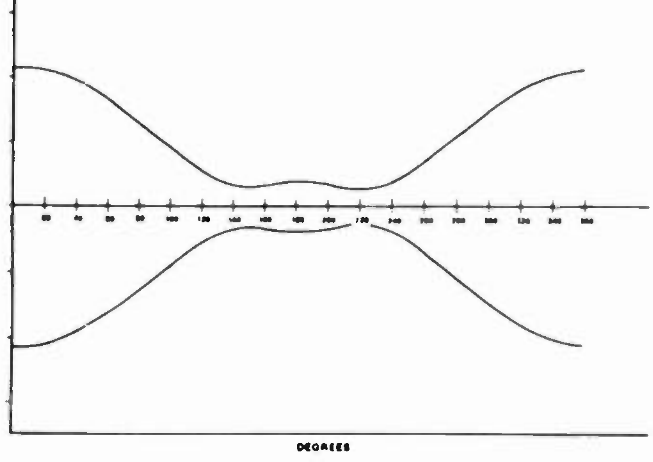


FIG. 10 ENVELOPE AT TRANSMITTER - DESIGN #2

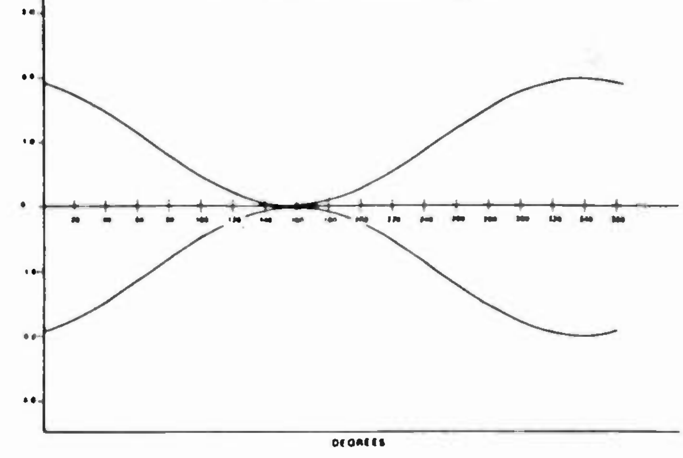


FIG. 8 THD - DESIGN #1

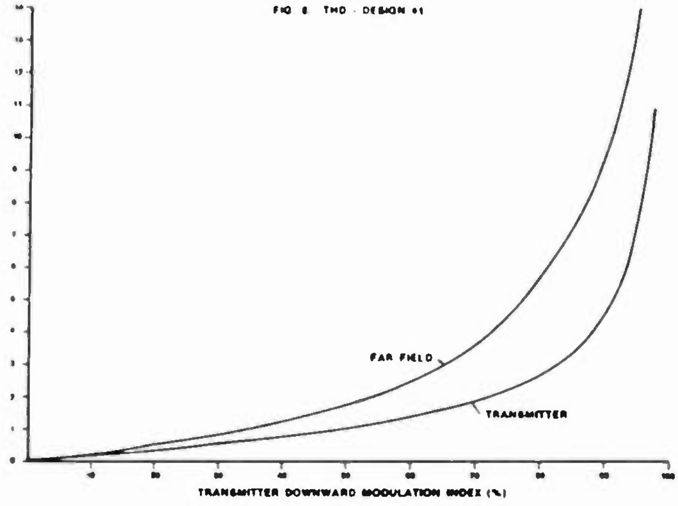


FIG. 11 ENVELOPE AT FAR FIELD DESIGN #2

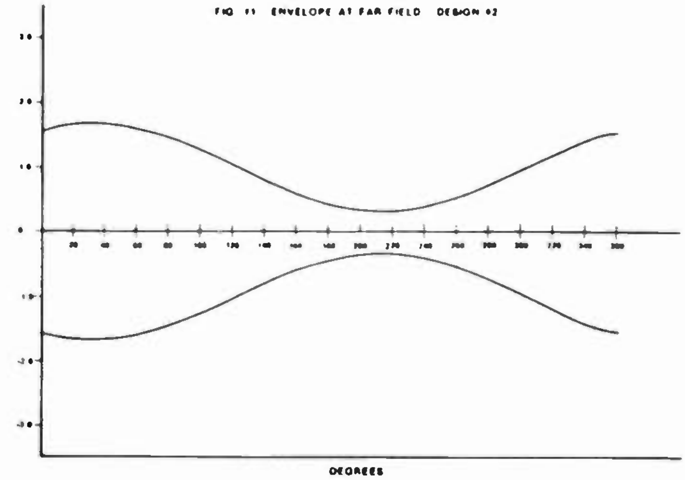


FIG. 9. MODULATION INDEX COMPARISON - DESIGN #1

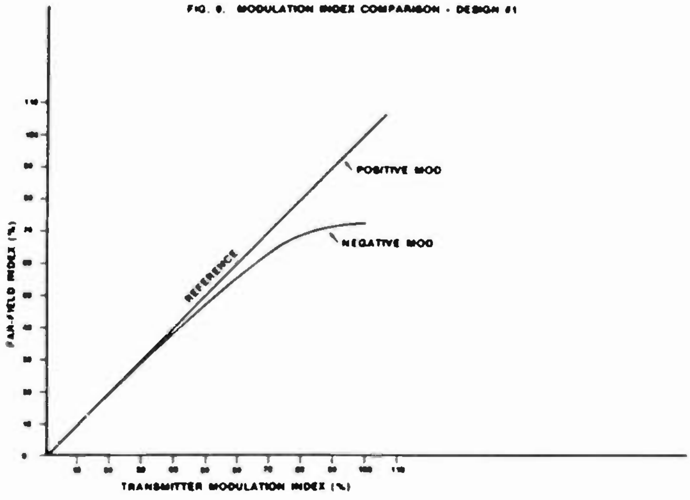


FIG. 12 MODULATION INDEX COMPARISON - DESIGN #2

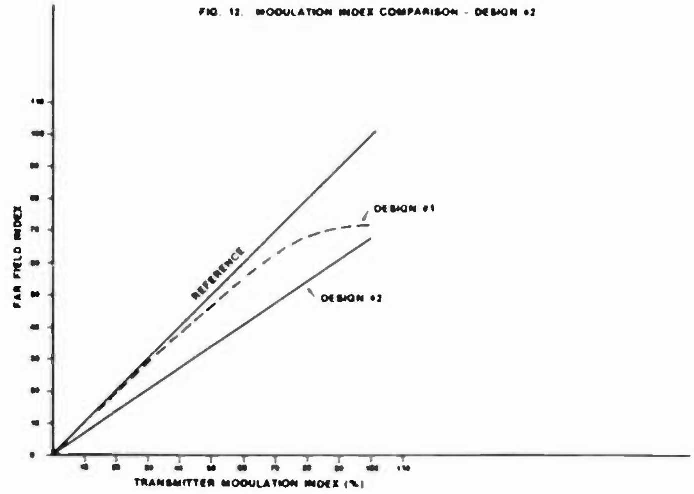


FIG. 13 MODULATION INDEX COMPARISON - DESIGN #3

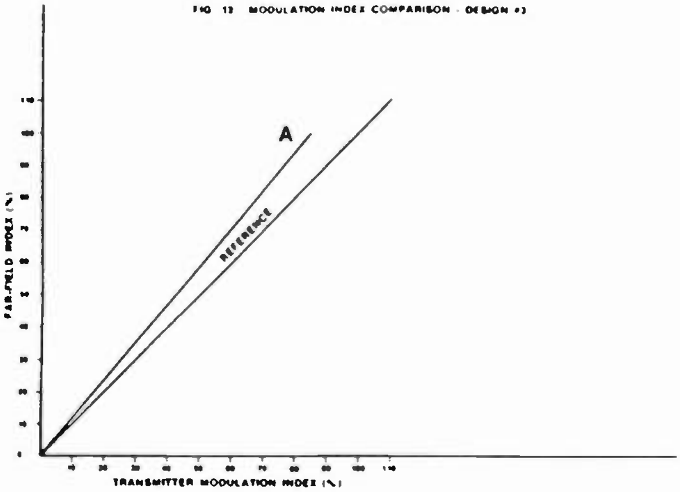
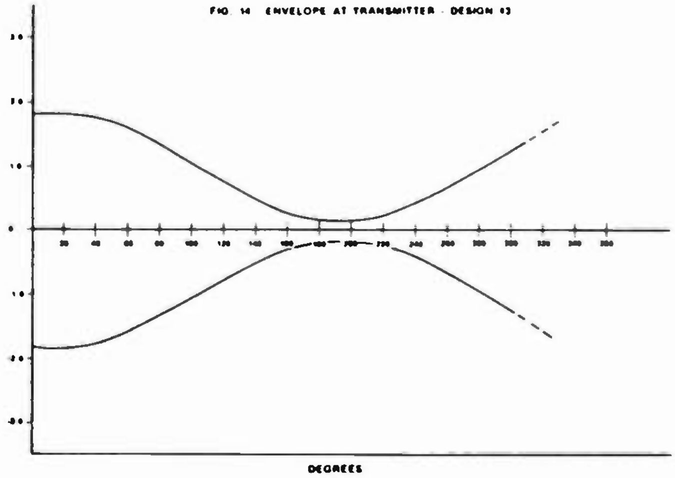


FIG. 14 ENVELOPE AT TRANSMITTER - DESIGN #3



AMPLITUDE MODULATION THEORY AND MEASUREMENTS — NEW AND OLD PARADOXES

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INTRODUCTION

Most electrical engineers find modulation theory both challenging and satisfying. The mathematics that services this branch of communications engineering covers a wide span, from elementary trigonometry to the finer points of Fourier analysis and Bessel functions theory.

Furthermore, aspects of sideband analysis that are now taught as being "obvious", (a most dangerous word, especially in patent law) were hardly obvious in the early days of radio engineering when even the inventors researching the field often "hedged their bets" by describing their inventions two ways; one in the time domain and the second in the frequency domain, in terms of the "new" sideband theory.

Indeed, it is reported by M. G. Scroggie in his informative book "Essays in Electronics" 1963, Illiffee Books, London, that in the 1930's the question as to the reality of sidebands was so heatedly resisted by a number of prestigious engineers, (including Sir Ambrose Fleming, the inventor of the vacuum tube diode) that the government of Great Britain appointed a committee to investigate. This committee came to the correct finding, sidebands were real!

The present paper was written in an attempt to serve two purposes:

(1) To dispel certain apparent misconceptions concerning modulation theory that have found their way into our profession, many starting at the very inception of communication engineering and others introduced as recently as 1986. Some of these apparent misconceptions may impact on efforts to update AM radio. In view of all the problems now facing AM radio, it would be suicidal for the AM industry to make major changes in its technical plant based upon fallacious or misleading technical concepts.

(2) Review methods of measuring "splatter", a most serious form of adjacent channel interference. (Splatter is a transient form of interference caused by sideband components falling into an adjacent channel.) While this review is extremely brief, it does discuss a basic limitation to such measurements that severely impacts on the validity of some of the measurement methods now being considered.

Basic Trigonometric Equation Defining an AM Signal

For single tone modulation;

$$e = Ec\left[1+m*\cos(wat)\right]*\cos(wct)$$

$$= \underset{\text{carrier}}{Ec*\cos(wct)} + \underset{\text{upper sideband}}{m/2\ Ec*\cos(wct+wat)}$$

$$+ \underset{\text{lower sideband}}{m/2\ Ec*\cos(wct-wat)}$$

where;

Ec = the carrier voltage
m = the modulation factor (1=100%)
wa = 2*pi*fa, fa is the audio tone frequency
wc = 2*pi*fc, fc is the carrier frequency

Thus, we see that for a single tone wave 100% (m=1.0) modulated AM wave, with a peak envelope voltage of 2 volts (4 watts PEP if across a 1 ohm resistor) there are three components a carrier, center frequency component of 1 volt, a lower sideband component of .5 volts and a mirror symmetrical upper sideband component of .5 volts.

NEW AND OLD MODULATION PARADOXES

Basic Modulation Paradox

The first point that confounded early researchers, and caused some to believe a sideband was merely a "mathematical fiction", is the difficulty of accepting the concept that a combination of three constant amplitude, constant frequency waves produces a single wave having a variable amplitude envelope. By use of the phasor diagram of FIG. 1(c) and armed with 20/20 hindsight, the original doubts of the reality of the mathematical defining equations are quickly dispelled. However, as mentioned above, all of this was not so readily accepted in the early days of radio. The equivalent paradox for FM, is how can a large number of components (actually infinite) all having, for fixed amount of deviation, constant amplitudes and constant frequencies combine to produce a single wave that varies in frequency.

The second point (not really a paradox but a bit of confusion that arose, I understand, in some schools) is that one can "see" sidebands when viewing a modulated wave in the time domain; i.e., a normal time sweep oscilloscope display.

This concept springs from an incorrect interpretation of a time domain representation of an AM wave (FIG. 1(b)) where some engineers believed the top of the figure represents the upper sideband and the bottom the lower sideband. However, it is impossible to see sidebands with this display, which is not frequency sensitive. To "see" sidebands you need a frequency domain display such as provided by a spectrum analyzer.

[Of course, if the modulated wave is fully defined in the time domain by a descriptive equation, or by a sufficient number of sample points spaced in time, one can determine the magnitude and phase of the sidebands and the carrier by performing a Fourier analysis.] A corollary of this erroneous concept; that a diode detector will demodulate one sideband if it is connected one way and the other sideband if the diode's connections are reversed, also has no factual basis. The diode merely responds to the envelope of the amplitude modulated wave which is defined by the summation of the carrier and both sidebands.

Overmodulation, Is It Really a Major Source of Interference?

NAB published, in September of 1986, a lengthy document as part of its AM improvement program, ("Modulation, Overmodulation, and Occupied Bandwidth: Recommendations for the AM Broadcast Industry" by Harrison J. Klein.) The document provides an extensive discussion of amplitude modulation theory with emphasis on the subject of "Splatter".

The main purpose of that document was to provide practicing broadcast engineers information that would better allow them to reassess their operating methods so as to reduce interference to other stations.

It is clear, that if AM radio is to fully compete with FM, receiver manufacturers must dramatically improve the fidelity characteristics of their products.* However, as is made clear in the NAB publication, wideband receivers are more prone to respond to adjacent channel interference. Therefore, the trade that AM broadcasters must be willing to accept is clean spectrum for wideband receivers. Anyone who believes in the long term future of AM broadcasting; therefore, must not ignore sources of adjacent channel interference, which include excessive preemphasis, high degrees of clipping to provide loudness, incidental phase modulation, poorly maintained transmitters that have high degrees of distortion and, not to raise a delicate issue, certain types of AM Stereo operation, where the L-R components can negate limits placed on the L and R input signals.

*For a thorough analysis see U.S. Department of Commerce publication dated Feb. 1987, "AM Stereo and the Future of AM Radio", Alfred C. Sikes, Assistant Secretary of Commerce.

The question spotlighted by the NAB publication is whether "overmodulation" is a significant factor in causing interference.

Unfortunately, the NAB publication reached certain conclusions by applying the results of an analysis of a wave generated by a very special class of amplitude modulator; i.e., a multiplicative type, for example, a balanced modulator. This line of reasoning lead to statements incorporated in this lengthy document that seriously downplayed the danger of overmodulation.

For example, the first conclusion in the "Executive Summary" of this NAB document, subtitled "Recommendations for the AM Broadcast Industry" is:

> "1. The primary cause of splatter interference is not the disappearance of the carrier during overmodulation, but instead is the presence of excessive high-frequency content in the audio signal that modulates the transmitter."

Furthermore, later in the document in the "Conclusions and Recommendations" it is stated:

> "1. The primary cause of splatter is excessive high frequency content in modulating audio. This is the single most important conclusion to be drawn from this report. If the modulation contains excessive high frequencies, these will cause splatter. Traditional "overmodulation", meaning the pinchoff of the carrier, is undesirable, but is much less significant than the audio itself. Carrier disappearance, commonly believed to be the significant cause of splatter, has little to do with it."

It is the present author's opinion that these statements tend to be misleading as it is impossible to operate a conventional AM transmitter so as to suffer "carrier disappearance" without causing severe interference.

It should be emphasized that the basic weakness in the analysis is quite subtle and the above quoted conclusions appear to be logical conclusions derived from an examination of a very special case.

Paradox That Almost Destroyed FM

This situation proves, once again, that modulation is a most arcane subject that can confound even the best trained engineers. Indeed, the prestigious engineer and mathematician, John R. Carson of Bell Laboratories, in 1922 published an analysis of narrow band FM that so biased the industry against FM that it delayed the acceptance of Major Edwin H. Armstrong's monumental development of wideband FM as announced in 1936. The early Carson paper concluded; "This type of modulation inherently distorts without any compensating advantages whatever."

It is noteworthy that another giant in radio engineering, F. E. Terman, stated in the 1937 edition of the classic, "Radio Engineering":

> "Frequency modulation is not particularly satisfactory as a means of transmitting intelligence. The frequency band is at least as great as that employed with amplitude modulation and is in general somewhat greater. Also the reception of frequency-modulated signals is not so simple a matter as the reception of amplitude-modulated waves."

This illustrates the danger of making broad generalizations (all FM signals are useless) based on special case examples, (analysis of narrowband FM).

Two Types of Overmodulation

Let us now follow the logic of the NAB publication. The balanced modulator theoretically produces a distortion free double-sideband suppressed carrier wave. If one wishes to produce a wave that, for modulation of less than -100%, i.e., no overmodulation, is equal to a conventional amplitude modulation, it is only necessary to add a carrier component, having an amplitude equal to or greater than the sum of the two sidebands when the sidebands are at their maximum negative going instant. The carrier component must be directly in line with the sum of the two sidebands if a pure amplitude modulated wave is to result.

If such a generator is caused to overmodulate, (by either upping the audio fed to the balanced modulator or conversely reducing the carrier level) the output does not cut off or flatten off as would be true of conventional AM transmitters, FIG. 2(a), but "folds over", FIG. 2(b).

This "fold over" overmodulation does not, as the NAB paper correctly points out, cause the sudden generation of higher order spectrum components. It does cause however, severe distortion in an envelope demodulator but not in a product demodulator. Actually, the "fold over" envelope distortion is over twice as high (23.9%) as is normal overmodulation clipping distortion (10.8%) at +141% modulation.

Thus, the NAB publication reasons overmodulation, per se, is not a source of excessive splatter. It is also true that the cusps (the sharp points where the envelope just goes in and out of the fold over region) are discontinuities but they also do not create splatter.

Another Paradox

Thus, we face yet another paradox. Years of operating experience have proven to broadcast engineers that overmodulation does indeed cause severe out-of-band radition, but we now see that a simple balanced modulator IC can produce a modulated wave capable of enormous amounts of overmodulation while maintaining excellent spectral characteristics.

The answer to the puzzle is that the overmodulated multiplicative wave is a lot more complex than one might conclude by viewing it on either a spectrum analyzer or an oscilloscope. The wave has another modulation component that is being ignored. The phase of the overall wave reverses at the very instant that the envelope folds over. This phase modulation makes the wave a very special case.

Actually the phase modulation of this wave produces a wideband spectrum as does the envelope modulation component. Indeed, the spectrum lines of both components of the overmodulated wave only approach zero amplitude as the sideband order approaches infinity! Thus, both the PM component and the AM components

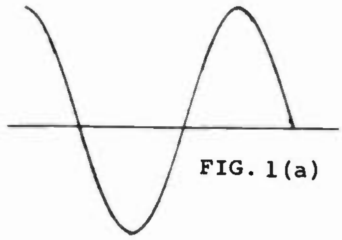

FIG. 1(a)

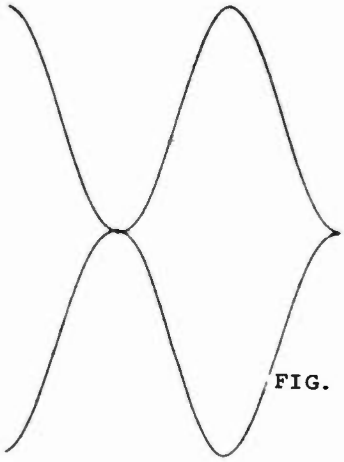

FIG. 1(b)

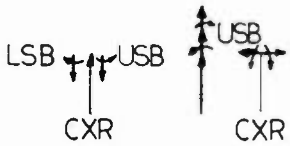

FIG. 1(c)

FIG. 1 SINGLE TONE 100% MODULATED AM WAVE

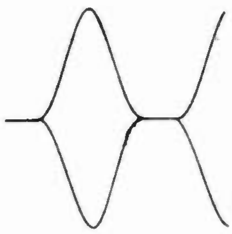

FIG. 2(a)

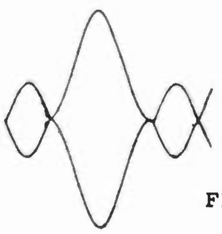

FIG. 2(b)

FIG. 2 TWO TYPES OF OVERMODULATION

are wideband waves whenever the overmodulation condition is reached; i.e., when "fold over"occurs.

This situation becomes clear if one compares FIG. 3 and FIG. 4.

In FIG. 3 we see a wave that would be produced by a conventional modulator if it was fed an AF wave with a folded over region and the wave just went through -100% modulation at the cusps. The spectrum is exceedingly poor....actually a good deal wider than a normal "flat bottomed" overmodulated wave.

On the other hand, if we use a balanced modulator type modulator, we will produce the identical envelope wave shape as shown in FIG. 3 but with the rf phase reversed in the fold over region (FIG.4). This phase reversal is sufficient to develop a wideband component that, when envelope modulated by the folded over envelope wave, produces a theoretically perfect double sideband spectrum.

Thus, we see that the wideband PM wave component is essential if this wave is not to splatter when overmodulated. In other words, the deduction that AM splatter is not caused by overmodulation is false, and the wave that was used to deduce this erroneous conclusion was actually not a pure AM wave but was a hybrid modulated wave with its PM component having the exact wideband characteristic necessary to cause a cancellation of the splatter components that would have been produced by the overmodulated envelope modulation.

Actually, without the PM component the "folded over" AM wave (FIG. 3) has a spectrum that violates FCC rules even when the audio frequency is restricted to voice frequency components below 3 kHz. In other words, as most broadcasters would expect, heavy overmodulation can cause excessive interference, even when transmitting low fidelity talk programs.

Actually, one can (and should) make a strong statement concerning overmodulation. Overmodulation, of a conventional AM transmitter, is a most serious source of splatter. Indeed, under normal conditions, overmodulation may well be the main cause of adjacent channel splatter.

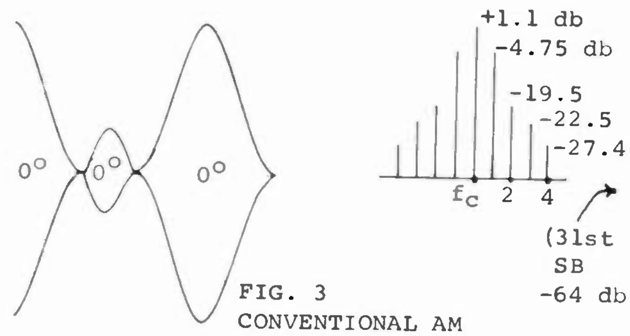

FIG. 3
CONVENTIONAL AM

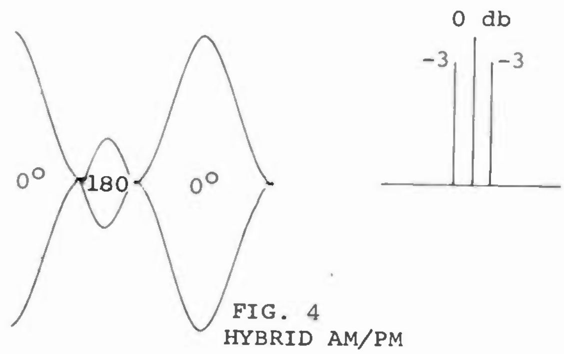

FIG. 4
HYBRID AM/PM

MEASURING SPLATTER

With the advent of AM Stereo there have been a number of proposals to replace the single tone test procedure for proving compliance with occupied bandwidth rules. The author believes that any group recommending the replacement of a standard test procedure must expect to be treated with some degree of suspicion and bears the burden of proving, beyond any question, that the proposed new method is better than the existing test procedure and that it provides at least as much information about the equipment being tested.

Furthermore, since broadcasters and receiver manufacturers agree that adjacent channel splatter is one of the most serious problems facing AM radio, it is even more important that any new test procedure proposed to measure splatter be thoroughly investigated.

List of Test Methods

Standard Single Tone	Sweep of audio from 50 to 15,000 Hz at 25% to at least 75%, preferably to maximum modulation capability of transmitter.
Multitone	For example, test proposed by NAMSRC 4 Tones; 400 Hz (35%), 2500 Hz (25%), 5500 Hz (15%) and 9500 Hz (10%).
Noise	White noise.
Shaped Noise	White noise passed through filter which supposedly simulates the characteristics of program material.
Normal Programming	Whatever station is transmitting during test period or "typical" programming.
Special Programming	Selected program material that hopefully stresses the equipment being tested.

All of the above test methods, with the exception of the standard single tone test as used by engineers since the beginning of radio engineering, present serious measurement problems. The basic problem with these alternative measurement procedures becomes evident when one considers certain fundamental measurement principles.

Some Measurements Are Inherently Inaccurate

At the outset it is worthwhile to consider the fact that measuring the frequency and amplitude of a sine wave is, and can only be, an approximation.

This concept becomes clear when one recognizes that a sine wave is not physically reliable as it is supposed to have started at minus infinity in time and must continue forever into the future. If you generate a sine wave in the laboratory you are only generating an infinitesimal part of a true sine wave. Therefore, when you turn "on" a tone generator, you generate keying sidebands, just as you do when you turn the oscillator off at night.

Fortunately, for the communications engineering profession, the bandwidth of these pseudo sine waves becomes extremely narrow in just a few seconds, allowing one to use a very narrow filter to determine the frequency of the sine wave. (Using a frequency counter does not change the situation in that you have to give the counter time to determine the frequency - the more precise the measurement, the longer the count.) Such limitations are met in other fields and apply to the accuracy of atomic observations (see"Theoretical Physics" by Georg Joos - Blacke and Son Ltd., Glasgow, 3rd Edition 1958) where Heisenberg's Uncertainty Principle established the fact that certain combined measurements of atomic particles are not only technically difficult but inherently impossible. In the instant case, because splatter is a very short term phenomenon, precise measurements are impossible.

Thus, attempts at determining splatter characteristics by use of narrowband spectrum analyzers are inherently inaccurate. Actually, the shorter the time span of the phenomenon, the less precise will be the frequency measurement.

On the other hand, steady state single tone measurements are quite accurate and repeatable, allowing regulatory agencies to be certain that licensed stations meet FCC rules.

Furthermore, peaky noise or program material can cause observers to see readings which will always tend to be low. Such problems are even more significant when conventional spectrum analyzers are used because:

1) The splatter component may not be present at the instant the analyzer's narrowband filter sweeps through the component's frequency.

2) The rise time of the analyzer's narrowband filter may be too long (too slow) to respond.

3) The phosphor's rise time of conventional display devices may be too slow to provide full response for faster transient components.

4) The viewer's response time may be too long to see short transient phenomenon and lack of concentration may cause the viewer to miss components.

Some of these difficulties can be alleviated by using a special spectrum analyzer with digital storage or by photographing or video recording the display information.

But even if such equipment is available, its use cannot overcome the basic inaccuracy problems due to the physics of measurements. Even "real time" analyzers, which use a multiplicity of stationary narrowband filters, while overcoming sweeping problems are still limited by the response time of the filters.

Single Tone Tests, Tough But Accurate

On the other hand, conventional single tone tests,while denigrated by some engineers (possibly because they have problems in passing such tests) have basic advantages such as, simplicity, low cost, high accuracy and repeatability. This does not mean that special noise tests and even mere listening tests are not useful. Indeed, it is inconceivable that a competent engineer would ever use a new broadcast technique without first carefully listening to the new technique with typical (and even atypical) radios. Listening to adjacent channels, especially if the transmission systems are A/B switched, can be quite revealing.

Especially questionable are shaped noise tests based on "average" spectrum measurements of voice and music. Industry experience with the standard FM preemphasis curve has shown the peak amplitude of short term high frequency components has been seriously underrated. With the advent of digital recording and electronic music it should be clear that the statistics of music are complex and few engineers are foolhardy enough to jump to conclusions based upon such statistics.

The author would like to note that he is not opposed to expanding test procedures as long as the basic single tone test is maintained as part of the overall test plan. Indeed, having done early work on shaped noise testing of new modulation techniques, he recognizes the fact that such techniques can be useful, especially if instead of using standard noise generators, repeatable computer generated pseudo-noise waves are used.

The important point is that there is no valid reason for discontinuing standard single tone testing. The situation is analogous to the continued use of harmonic distortion testing even though intermodulation distortion measurements have been, for decades, thoroughly accepted.

CONCLUSIONS

Overmodulation Causes Splatter

In view of the importance of correcting what the author believes is potentially a serious source of misunderstanding that may cause some stations to ignore the dangers of overmodulation, the author wishes to make the following statement as strongly as possible; i.e., there is no practice that can, under normal conditions, cause more adjacent channel interference than overmodulation of a typical AM transmitter.

When the level of the audio signal, fed to an AM transmitter, is increased past a point where the transmitter's output will not decrease, the problems of sharp negative peak clipping are experienced. Indeed, the effect is always worse than an equal amount of negative clipping in the audio processor feeding the transmitter because, except for the output rf tuning and bandwidth limits of the antenna, there is no filtering.

Furthermore, the worst problems of IPM (Incidental Phase Modulation) are generally experienced when a transmitter is forced to approach the region of 100% negative modulation.

Thus, the main purpose of this paper is to dispel the slightest doubt that "sharp edges" and overmodulation of AM transmitter signals have a tremendous effect on splatter, something most broadcast engineers knew all along. This is one paradox that cannot be allowed to confuse the industry for a long period of time.

Single Tone Testing Must be Enforced to Protect Spectrum

Regarding new techniques for measuring splatter, such techniques are to be encouraged especially if they can help guard against new and unusual sources splatter and other forms of interference. However, until rigorous proofs are available guaranteeing that such new techniques cover all aspects of present test practice, both the new and old test procedures must be used if the spectrum is to be properly protected. It is abundantly clear that the AM radio industry cannot accommodate even the slightest increase in interference beyond present rules.

It is certainly not the time to abandon tried and proven test procedures, just because these tests are difficult to pass.

ADVANCED TELEVISION SYSTEMS

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ABSTRACT

This NAB Workshop paper was given as an FCC Tutorial in January, 1987 to acquaint FCC engineers with progress in advanced television systems. It is repeated here for the same purpose for the broadcasting community. This paper reviews the NTSC system and possible improvements to it, component systems, the 1125/60 HDTV studio system, and various proposals which have been made for transmitting high definition television to the public.

BACKGROUND

The 525 lines per frame, 60 fields per second, 2:1 interlaced television system has been serving the United States public for almost 50 years. Performance of the system has improved significantly over the years, clearly one of the reasons for its long life.

The most significant single improvement was the addition of color. Engineers were able to add color information to the black and white signal without increasing the transmission bandwidth. To achieve this the luminance information was decreased and a subcarrier, containing the color information, was introduced. The result for black and white receivers was lower resolution and the appearance of a dot structure, a loss that was considered to be acceptable.

Other improvements have taken many forms and arise from constantly expanding technology. Both pick-up devices and display devices have improved dramatically. Solid state circuits perform complex functions which were not possible when the system was designed.

Technology now permits another significant improvement, high definition television. Although everyone recognizes high definition television is here to stay, there have been many debates on the need for high definition television, the precise timing of various services and, of course, the technical standards.

This paper will first review the NTSC system, pointing out some of the artifacts. Various improvements that can be made to the NTSC system will be examined.

A review of multiplexed analog components (MAC) will then follow. MAC systems have been proposed for DBS services in several parts of the world. Since some of the proposed HDTV transmission systems use MAC technology, it is appropriate to establish an understanding of MAC systems before covering the proposed HDTV transmission systems. The 525/60 B-MAC system will be reviewed in this section.

The next topic will be the 1125 lines per frame, 60 fields per second, 2:1 interlaced high definition television studio system, the only HDTV studio system that has been designed and reduced to practice. Many of the transmission proposals assume a high definition studio signal will exist and are designed with the studio system as an input to the transmission system. The world-wide standards effort in the CCIR has concentrated on a studio standard. This therefore seems to be an appropriate starting point for review of HDTV transmission systems.

Transmission of high definition television is the final topic. The general concepts will be noted followed by descriptions of specific proposals. The similarities and differences of the various proposals are given.

The reference materials used to prepare this paper are listed by topic at the end of the text. For those wishing to study advanced television systems in greater detail, these papers, in addition to the references cited in the individual papers, would constitute a remarkable library.

NTSC

Although many people are not aware of this fact, there have been two NTSC's. The first National Television System Committee was convened around 1940 to establish the technical standards for an American black and white television system. The agreed upon standards were 525 lines per frame, 60 fields per second, 2:1 interlaced and 4:3 aspect ratio. The field frequency was precisely 60 Hertz. The sound carrier frequency was set 4.5 MHz above the picture carrier frequency. The maximum video bandwidth transmitted was 4.2 MHz. Channel spacing was 6.0 MHz.

The second NTSC was convened in the early 1950's to establish technical standards for an American color television system. Black and white parameters were maintained with one exception which will be noted later.

The color information was added to the black and white signal by inserting a subcarrier modulated in quadrature by two color difference signals. The two color difference signals, called the I and Q signals, are in quadrature on a color diagram. The I signal was specified with a bandwidth of about 1.5 MHz while the Q signal specification was only .5 MHz. The human eye has greater color resolution for colors near the I axis than near the Q axis and thus, to conserve bandwidth, these axes were chosen. The equations for the luminance signal (Y) and the color difference signals are derived from the red, green and blue signals as follows:

$$Y = 0.30*R + 0.59*G + 0.11*B$$

$$I = -0.27*(B-Y) + 0.74*(R-Y)$$

$$Q = 0.41*(B-Y) + 0.48*(R-Y)$$

The color subcarrier frequency was chosen to be an odd multiple of one half the horizontal scanning frequency to minimize the appearance of the subcarrier in the picture. The multiple was also selected to have small factors. The resulting relationship is given by:

$$f_{SC} = 13*7*5/2\ f_H = 455/2\ f_H$$

Since there were concerns that the sound carrier frequency could not be changed (and maintain compatibility with receivers already in use) and that the color subcarrier and sound carrier would cause mutual interference, it was decided that the horizontal scanning frequency (and thus the field frequency) would have to be changed. To minimize the interference the horizontal scanning frequency was interleaved with the sound carrier. The ratio of the sound carrier frequency and the horizontal scanning frequency had been 4,500,000/15,750=285.71. The horizontal scanning frequency was changed so that the sound carrier frequency would be an even multiple of one half the horizontal scanning frequency. The factor closest to 285.71 satisfying this requirement, 286, was selected. This made the new field frequency precisely 1000/1001 times 60 Hertz, thus giving rise to the mysterious 59.94 Hertz vertical scanning frequency. Figure 1 is a block diagram of an NTSC encoder. The marked points A and B will be referenced when the subject of precombing is introduced.

The manner in which the color information was added gives rise to some of the artifacts observed in the NTSC system. High spatial frequencies in the imaged scene can produce luminance information which is treated by the NTSC decoder as if it were color information. Also, a wide bandwidth luminance channel in a receiver can cause the color subcarrier to be displayed as a dot structure. In each case the artifact arises from the mutual interference of the luminance signal and the color signal.

Another artifact arises from interlaced scanning. The raster appears to slowly move up the screen and, once the human eye has locked onto this movement, the resolution of the picture appears to be lower. The scanning structure becomes obvious. If the viewer's eye follows objects in motion in the displayed image, again, the resolution of the picture appears to be lower and the scanning structure becomes obvious.

IMPROVED NTSC

The most promising concepts for improving NTSC are:

1) progressive scanning in the display,

2) progressive scanning in both the camera and display maintaining the interlaced transmission,

3) pre-combing luminance and chrominance prior to transmission and

4) the Fukinuki proposal which sacrifices a small amount of color information to increase the luminance information.

Progressive Scan in Display

Current television systems use 2:1 interlace scanning. Two vertical scans, or fields, are required to complete one frame of the picture. Each field contains half the total number of scanning lines. The first field provides every other line of the frame; the second field contains the other set of every other line. Figure 2 illustrates the 525 line 2:1 interlaced scanning raster.

A progressive scan system (also called sequential scan) contains all the scanning lines in each field. Scanning 525 lines each field produces a better picture than interlaced scanning but it doubles the bandwidth.

The picture resulting from an interlaced signal can be improved by converting the interlaced signal into a progressive scan signal with two times the number of scan lines. The scan line visibility decreases by a factor of two. This technique does not increase the signal resolution but it does increase the perceived resolution by improving the Kell Factor. The easiest way to implement this improvement is to display every scan line twice during the normal scan line period. One line store is required and the horizontal scanning frequency is doubled. Diagonal lines in the picture become distorted.

A further improvement, requiring two line stores, inserts the average of two time-adjacent scan lines between the said two lines. The distortion of diagonal lines is decreased.

Both of these techniques are effectively creating a new scan line in between the existing scan lines. Figure 2 illustrates this point. Note that there is already a scan line in this location, the scan line in the other field. By adding a field store this line in the other field can be displayed. This process results in very good still pictures. However, it also produces motion defects since the line in the other field is separated in time by 1/60 of a second. Motion compensation circuitry is required for best results.

Progressive Scan in Camera/Display with Interlaced Transmission

Still greater picture improvement can be obtained if the camera uses progressive scan and the signal is converted to an interlaced signal prior to transmission. In this case extra information is available at the transmitting end to process the transmitted lines in such a way that, when the receiver re-converts the signal to progressive scan, the final picture will be improved. Even greater benefits could be obtained if auxiliary information were transmitted to tell the receiver a little more about the best way to put the picture together again. This approach may create a problem for normal receivers by transmitting higher vertical resolution than can be displayed and therefore causing aliasing. It is not clear whether this approach would require new standards.

This technique does not require one more standard for studio cameras. The high definition television studio signal can be an input to this system with the signal scan converted to 525/60/2:1 anticipating progressive scan in the receiver.

This approach, more scan lines in the camera and in the display with interlaced transmission, has been proposed by Bell Labs and Philips to increase vertical resolution in high definition transmission systems.

Pre-Combing

An artifact quite visible in the NTSC system results from interference between higher resolution luminance and the color signal. This occurs because of the overlapping spectra. If the luminance and color difference signals are filtered to eliminate overlapping spectra prior to NTSC encoding, these artifacts are greatly diminished.

Figure 3 illustrates the phase of the color subcarrier on successive scanning lines. The phase shift from one line to the next line is exactly 180° because the subcarrier frequency is an odd multiple of half the horizontal scanning frequency. This characteristic can be used in special filters, called comb filters, to separate luminance and color information. If the signal is in phase from one line to the next line it is assumed to be luminance information. If the signal is out of phase from one line to the next line it is assumed to be color information.

Figure 4 is a block diagram of a comb filter. The configuration shown will pass only luminance information, rejecting the color information. The resulting spectrum is shown in Figure 5. If the signal between the delay lines is inverted prior to the adder, only the color information passes through the filter. The resulting spectrum is identical to that shown in Figure 5 except that the nulls, rather than the peaks, occur at multiples of the horizontal scanning frequency.

Figure 6 illustrates the circuitry which must be added at points A and B in the block diagram of Figure 1 to pre-comb the luminance and chrominance. Adding this circuitry to every camera would not be a good practice, however, since the current practice in studios is to use multiple NTSC encoders and decoders (to perform digital video effects, for example). Such an improvement may be more appropriate in a component studio in which NTSC encoding is done one time only, immediately prior to transmission.

Proponents of this improvement claim that significant improvements are visible in receivers with comb filter decoders. They further claim, surprisingly, that some improvements are seen with traditional receivers.

Fukinuki Proposal

Dr. Fukinuki, Hitachi Central Research Laboratory, proposes interleaving higher definition luminance with color in much the same way as the color information is already interleaved with the luminance information. He points out that a portion of the spectrum devoted to color is poorly used and this portion could be dedicated to high resolution luminance information. This technique produces motion defects and thus motion compensation circuitry must be included in the receiver.

Figure 7 shows the television signal in a three dimensional representation with the horizontal scanning lines parallel to the x-axis, the vertical scans parallel to the y-axis, and time the third dimension. In this case, each field, separated in time by 1/60 second, represents a plane. Just as the color subcarrier has a 180° phase shift from line to line as illustrated in Figure 3, it also has a 180° phase shift from field to field and frame to frame. This characteristic can be used to carry color information with one phase shift and high resolution luminance information with another phase shift. If the signal is in phase following a 262 line delay, it is assumed to be color information as shown by the dashed lines rising to the right in Figure 7. If the signal is in phase after a 263 line delay, it is assumed to be high resolution luminance information as shown by the dashed lines falling to the right.

Figure 8 is a block diagram of the decoder proposed by Dr. Fukinuki. Signals out of phase after a 262 line delay are decoded as high resolution information. Signals out of phase after a 263 line delay are decoded as color information.

MAC SYSTEMS

Several different MAC systems have been proposed in the standards efforts around the world. Their similarities are great; the differences are in the precise choice of numbers. The 525/60/2:1 B-MAC system will be illustrated in this paper.

Before proceeding it would be helpful to note that television was developed mostly as a service to the public. Engineers reached agreement on the transmission parameters. Broadcasters then used these parameters to make television programs. It was convenient, perhaps mandatory, to use the transmitted format as the studio production format. Separate components were difficult to use because of timing constraints. As technology has advanced (consider digital video effects) the need for higher performance in the studio has increased. Sufficient headroom should exist for full transmission quality after all post-production.

MAC systems came about as a convenient way to maintain separate components without having to worry about maintaining the critical timing of three separate signals on three separate cables. Figure 9 shows the B-MAC waveform. The luminance and color difference and multiple digital sound signals are compressed in time and placed on the same signal line. Various MAC systems differ in their compression ratios, number of sound channels, and data rates. B-MAC compresses the luminance by the factor 3/2 and compresses the color difference signal by a factor of 3. The data format provides six high quality digital sound channels. The color signal uses a line sequential format. The R-Y and B-Y signals are carried on alternate lines.

The B-MAC system can accommodate the wider aspect ratio of 16:9. Consider first the compression/expansion technique used when a 4:3 aspect ratio signal is displayed. One scan line of the luminance signal, 750 samples, is placed in a line store with a 910 f_H clock. The samples are removed from the line store with a 1365 f_H clock, compressing the luminance signal by the factor 3/2.

At the decoder the 750 samples are placed in a line store with a 1365 f_H clock and removed from the line store for display with a 910 f_H clock. This process expands the luminance signal by the factor 3/2, restoring its original form.

Identical processes are used when a 16:9 aspect ratio signal is displayed on a 16:9 aspect ratio monitor. If, however, the 16:9 aspect ratio signal is displayed on a 4:3 aspect ratio monitor, the decoder must expand the luminance signal by the factor

$$(3/2) * (16/9) / (4/3) = 2$$

In this case, after the 750 samples are placed in a line store in the decoder with a 1365 f_H clock, the samples are removed from the line store for display with a 1365/2 f_H clock. The decoder does not use the entire vision signal. Pan and scan is accomplished by having the signal include information which tells the decoder which portion of the signal to display.

MAC systems can offer higher performance than composite systems because of the separation of the luminance and color difference signals, the inclusion of full bandwidth luminance, and the resulting lack of cross-modulation. On the negative side, the vertical resolution of the color information is lower because of the line sequential format. Also, the baseband bandwidth required for a MAC system is increased by the amount of the luminance compression ratio, 50% for B-MAC.

1125/60 HDTV STUDIO SYSTEM

NHK, the Japan Broadcasting Corporation, has been studying and developing HDTV for several years. Their scientists assumed that there were many applications of HDTV, including broadcasting. They conducted psychophysical experiments on the size of screen, the angle of vision, the sense of reality, etc. After the experiments were completed NHK designed a system which met the requirements, the 1125/60 HDTV system.

Many people around the world support this system for a single world-wide production standard. They argue that a single world-wide high definition electronic production standard is extremely desirable and that the 1125/60 system satisfies production requirements. Resolution is comparable to 35mm releases, a world standard for motion pictures.

Figure 10 shows the parameters generally agreed on for a high definition television studio using the 1125/60 parameters; NHK originally proposed the following basic parameters:

1125	lines per frame
60	frames per second
2:1	interlace
5:3	aspect ratio

The number of lines, 1125, was selected to be greater than 1000 but not twice 525 or 625, a compromise between the two scanning standards in existence today. They chose 60 frames per second, as opposed to 50 frames per second, because of the reduced flicker and the higher temporal sampling rate. They selected interlace scanning rather than progressive scanning because of the reduced bandwidth. They believed the aspect ratio should be at least 5:3, perhaps as wide as 2:1, and selected 5:3. Studies in the United States supported each of these parameters except the aspect ratio. The ATSC proposed that the aspect ratio should be 16:9 to give greater flexibility in shooting and releasing a program. By using a "shoot and protect" scheme with a 16:9 aspect ratio, releases could be made conveniently in any aspect ratio between 4:3 and 2.35:1. If the master has the 16:9 aspect ratio, a 4:3 aspect ratio release would use the full height of the master and the appropriate width as shown in Figure 11. A 2.35:1 aspect ratio release would use the full width of the master and the appropriate height, also illustrated in Figure 11. The outer rectangle represents the 16:9 aspect ratio master. The inner rectangle represents the image area in which the critical portions of the image should be contained.

Several engineers wanted a progressive scanning format to be used, arguing that post-production would be easier. However, with twice the number of lines per field, the bandwidth doubles. Camera sensitivity is reduced. Most engineers felt that the number of lines should not be decreased below 1000 to compensate for the greater bandwidth. One argument against progressive scanning was that, if the bandwidth were to be doubled, it would be preferable to continue to use interlaced scanning but with twice the number of lines.

NHK proposed that the studio system have separate luminance and color difference signals. This has general acceptance but the bandwidths being considered today are greater than those first proposed by NHK. The ATSC suggested that sampled representations of the signal should be specified as well as specific bandwidths. The EBU (European Broadcasting Union) suggested that the sampled representations, only, should be specified. In order to decide how many samples per line should be used, the ATSC argued that the CCIR has defined HDTV as having about twice the resolution, both horizontal and vertical, of current television systems. CCIR Recommendation 601 specifies 720 luminance samples during the active line and half that number for each of the two color difference signals for current television systems. Twice the resolution would then imply twice 720 samples multiplied by the ratio of aspect ratios (16:9 divided by 4:3) resulting in 1920 samples per active line for the luminance and half that number for each of the two color difference signals. The resulting bandwidths would be about 30 MHz for luminance and 15 MHz for each of the color difference signals.

Various numbers have been proposed for the specific sampling frequency (which would also specify horizontal blanking) but no single number stands out as the obvious choice. Many engineers prefer a short horizontal blanking period to lower the sampling frequency. Manufacturers worry that a short blanking period will increase the cost of receivers.

NHK also proposed that NTSC colorimetry be used for the HDTV studio system. There has been relatively little debate on this subject. The debates have centered on the field frequency.

NHK has recently proposed a new concept for the synchronizing signal, a three level signal as shown in Figure 12. The precise timing information is carried by the zero crossings between negative and positive pulses rather than the negative going edge of synchronizing signals used today. NHK believes the timing accuracy improves significantly with this waveform.

The ATSC agreed in March, 1985 to recommend to the U.S. Department of State that the 1125/60/2:1/16:9/1920 parameters be proposed to the CCIR as a single worldwide standard for HDTV studios. After the U.S. CCIR National Study Group unanimously agreed, the Department of State submitted this position to the CCIR as the U.S. position. The governments of Canada and Japan submitted similar positions. At the CCIR Plenary Assembly meeting in Dubrovnik in May, 1986 the decision on a studio standard was postponed until the next Study Period. The Plenary Assembly agreed unanimously to attach the parameters to the CCIR Report 801 making them the only parameters acknowledged in Report 801.

Since the time of the Plenary Assembly, activities around the world indicate that the 1125/60 system will probably become a de facto standard for 60 Hz HDTV studios. What is not clear is whether or not the system will be accepted as a single worldwide standard.

HDTV TRANSMISSION

Distribution of HDTV program material will likely be via consumer vcr, consumer video disc, optical/electrical cable systems, DBS transmission, and terrestrial transmission. I recognize that the last of these distribution media is probably the most difficult because of the standards and regulatory issues involved. However, it is my opinion that the terrestrial broadcasters, rather than roll over and play dead, will find a way (technical and/or political) to make significant improvements in the technical quality of their transmissions when the other distribution outlets begin using HDTV.

Will the HDTV technical standards for each of these media be the same? Although there may be advantages if they are the same, it is not clear that they must be the same. For example, bandwidth is most limited for terrestrial transmission and compromises will be necessary. In an analogy with audio systems, one would find that sound input devices (FM radio, AM radio, LP's, CD's, cassette recorders, reel to reel recorders, TV sound) vary widely but feed into a common amplifier and speakers. Perhaps the consumer HDTV system will consist of a display driven by a frame store with multiple inputs to the frame store (NTSC, HDTV-VCR, HDTV-UHF).

Compatibility is a term that is often used and too often misused. I propose that we define levels of compatibility related to receivers. The highest level (LEVEL 5) is represented by a system which allows HDTV transmissions to be received by a current receiver and displayed as an HDTV picture. Although this may seem absurd, the concept represents the highest attainable level of compatibility. The next lower level of compatibility (LEVEL 4) is represented by a system which allows HDTV transmissions to be received by a current receiver and displayed with the same quality as current transmissions. LEVEL 3 is represented by a system which allows HDTV transmissions to be received by a current receiver and displayed with reduced performance when compared with the picture from a normal transmission. This is what happened when the United States added color to the black and white television transmissions. The next lower level (LEVEL 2) is represented by a system which allows HDTV transmissions to be received and displayed by a current receiver using an inexpensive adapter box . This was the situation when UHF transmissions first began. The next lower level (LEVEL 1) is represented by a system which requires an expensive adapter box, perhaps so expensive that consumers would prefer to purchase the new system. In the cases of LEVEL 2 and LEVEL 1, I assume that a new receiver can be designed to operate on both the current system and the HDTV system. LEVEL 0, the lowest level and the only level which I would call non-compatible, is represented by a system with which current receivers cannot display HDTV transmissions in any form, even with adapter boxes, and new receivers cannot display current transmissions. The levels of compatibility are illustrated in Figure 13.

I believe that the highest performance HDTV transmission system will have a low level of compatibility. Also, I believe that high level compatibility systems will be lower performance HDTV systems. This factor must be recognized when making a decision. The trade-off involves today's level of compatibility versus tomorrow's level of performance.

Six proposals for HDTV transmission are examined in this paper.

1) MUSE Proposal

2) Bell Laboratories Proposal

3) CBS Proposal

4) Glenn Proposal

5) Del Rey Group Proposal

6) Philips Proposal

The proposals fall into three categories with respect to channel requirements.

1) A single channel wider than current channels

2) Two channels with one channel carrying a "compatible" signal

3) A single current channel which carries a "compatible" signal

The MUSE proposal requires one channel, wider than an NTSC channel, and is LEVEL 1 compatibility. The Bell Labs system uses two NTSC channels where one channel is LEVEL 3 compatibility and contains an NTSC signal. The CBS proposal is also a two channel system except it was proposed as a DBS system and uses a MAC approach, rather than NTSC, for the first channel and thus has LEVEL 2 compatibility with respect to NTSC receivers. The Glenn proposal uses one NTSC channel and another low bandwidth channel. The first channel contains NTSC and has LEVEL 3 compatibility. The Del Rey Group proposal requires only one NTSC channel and has LEVEL 3 compatibility.

Another system, the Philips proposal for 625/50 systems, will also be described. In the European context, where MAC DBS transmissions are anticipated, this system requires one channel and is described by proponents as being "compatible." This proposal has LEVEL 2 compatibility since the European MAC system is not operating today and the PAL and SECAM receivers in Europe would require adapters.

MUSE Proposal

Multiple Sub-Nyquist Encoding has been proposed by NHK for DBS transmission of HDTV. The signal is derived directly from the 1125/60 studio system. The luminance and color difference signals are band limited then sampled. One of every four samples is transmitted during each field and, after four fields, every sample is transmitted. This process, depicted in Figure 14, produces high quality still pictures. During each line 373 actual luminance samples are transmitted. The minimum horizontal spacing of samples is about 1/1500 of the picture width. The minimum vertical spacing of samples is about 1/1125 of the picture height. The resolution of objects in motion is lower than the resolution of stationary objects. Receivers require a frame store. Motion detectors are used in the encoder to fully compensate for some types of motion such as a camera pan. This information is transmitted to the receiver as a digital signal. Digital sound is transmitted. Luminance and color difference signals are kept separate in a MAC format. The full signal requires a baseband bandwidth of 8.1 MHz.

Although the MUSE system was designed for FM transmission, the MST-NAB demonstration (Washington DC, January 1987) used the MUSE system with vestigial sideband AM transmission and occupied two UHF channels (58 and 59). The picture carrier was set 3 MHz into the 12 MHz channel.

Several consumer electronics manufacturers in Japan are designing consumer equipment to operate with this system. Plans are being made in Japan for a DBS service, starting around 1990, using this system.

Bell Laboratories Proposal

Bell Labs proposed a two channel system in which the first channel contains an NTSC signal derived from a high definition signal with 1050 lines. The 1050 line signal, following vertical filtering, is scan converted into the 525 line format. Bell Labs claims this process increases vertical resolution by increasing the Kell Factor. The horizontal resolution of the signal transmitted in the first channel is normal NTSC. Higher frequency luminance and color difference information is in the second channel. Horizontal resolution of the combined signals is essentially two times NTSC resolution. Bell Labs claims an NTSC receiver recovers the single NTSC channel with only slight degradation. An HDTV receiver recovers both channels and combines the information in a frame store scan converting the output to 1050 lines producing a high definition picture.

Bell Labs claims the second channel has sufficient capacity to transmit multiple channel sound. They have also described several methods for obtaining wider aspect ratio pictures.

Figure 15 shows the transmitted spectrum. Although this figure shows two adjacent channels, two non-adjacent channels can be used. Figure 16 is a block diagram of the encoder. The decoder uses the inverse function.

CBS Proposal

CBS proposed a two channel transmission system for an HDTV service using two DBS channels. Each channel carries a time multiplex component (TMC) signal. In this paper, the TMC signal should be considered the same as a MAC signal. The CBS system first converts the HDTV studio signal into a 1050/60 format with a 5:3 aspect ratio. The first channel's signal is obtained by averaging each pair of lines of the 1050 line signal to create a 525/60 signal with 5:3 aspect ratio. As shown in Figure 17, the central 4:3 aspect ratio portion of this signal is transmitted in the first channel. The second channel carries every second line of the 1050 line signal in a 5:3 aspect ratio format. It also carries the "side panels" of the first channel.

An NTSC receiver, using an adapter box, uses the signal in the first channel to display a 525/60 picture with 4:3 aspect ratio. An HDTV receiver combines the two signals to display a 1050/60 picture with 5:3 aspect ratio.

Vertical filtering is applied to the first channel (averaging each two lines of the 1050 line signal) so that there will be no loss in the single channel receiver. The filtering is illustrated in a scanning format in Figure 18. The "side panels" have lower horizontal resolution than the central portion of the picture since they are transmitted in the second channel which is compressed by a greater factor. The horizontal spatial resolution of the resulting high definition signal is the same as the horizontal resolution of the signal in the first channel. The signal in the first channel, it should be noted, has about 50% higher horizontal resolution than an NTSC signal since the DBS channels permit transmission of a wider bandwidth signal. The vertical resolution of the high definition signal is two times the vertical resolution of an NTSC signal.

This system can be implemented without using a frame store in the receiver. The TMC format for each channel is illustrated in Figure 19.

Glenn Proposal

William E. Glenn of the New York Institute of Technology proposed a system using one NTSC signal and an auxiliary signal which occupies about one half an NTSC channel. Dr. Glenn made studies of human vision and found that humans have two types of vision receptors which have different functions for spatial resolution and for temporal resolution. His system takes advantage of these properties of human vision to reduce the transmitted bandwidth.

Dr. Glenn's NTSC signal is subjected to improvements using techniques described in the improved NTSC section of this paper. His auxiliary signal contains the higher frequency, lower temporal rate information which is used in a frame store to increase the resolution. The auxiliary signal also carries information to produce a wider aspect ratio picture. A block diagram of the encoder for this system is shown in Figure 20.

Del Rey Group Proposal

The Del Rey Group proposed a 525/60/2:1 high definition transmission system which uses a single NTSC channel. The sampling pattern is illustrated in Figure 21. The transmitted signal can be derived from an 1125/60 studio output. The easiest way to examine this proposal, though, is to look at an original luminance signal with twice 525 lines and three times the horizontal resolution. Each NTSC luminance pixel is replaced by six new pixels (three side by side with two rows up and down) as shown. The pixels designated A-F are transmitted in place of the normal luminance pixel, and, after six fields, all six pixels are transmitted. A frame store is used in the HDTV receiver to reclaim the full signal. The Del Rey Group claims that this signal could be directly displayed on a current NTSC receiver with little loss compared with a conventional NTSC picture. Normal NTSC color difference bandwidths are used in the system. The minimum spacing of horizontal samples is about 1/1300 of the picture width. The minimum spacing of vertical samples is about 1/1000 of the picture height.

The Del Rey Group also proposes that 69 fewer active video lines be transmitted each frame which results in a wider aspect ratio picture. The Del Rey Group claims that most NTSC receivers overscan to such an extent that the loss of the transmitted lines would not be observed in the typical receiver. Those lines are then used to transmit digital sound. Figure 22 shows this approach.

Philips Proposal

Philips has proposed a European "HD-MAC" transmission system using the MAC system with 625/50 interlaced transmission. A block diagram of this proposed system is shown in Figure 23. A 1250/50/2:1 studio system provides input signals. Vertical filtering is used to make a wide bandwidth 625/50/2:1 signal. Alternate horizontal samples are transmitted to reduce the bandwidth. Four fields are required for the HD-MAC receiver to obtain the complete signal. The display is 1250/50/2:1. A frame store is required in the receiver.

The sampling pattern is illustrated in Figure 24. The minimum spacing of the vertical samples is about 1/625 of the picture height. The vertical resolution is higher than this number implies, though, since the Kell Factor has been increased by the vertical filtering. The minimum spacing of horizontal samples is about 1/1400 of the picture width since 720 actual luminance samples are transmitted each line. The effective minimum spacing of horizontal samples, however, is about 1/1000 of the picture width because of filtering applied prior to the sampling process.

Similarities and Differences of Proposals

It is difficult to make direct comparisons between the proposals. Demonstrating the systems side by side using test signals and program material would provide the best comparison. However, this cannot be done today since only one of the systems has been thoroughly developed. The others are in various stages of development.

As a general rule, systems requiring the greatest bandwidth probably have the best performance. Likewise, systems using the least bandwidth probably have the poorest performance. Trade-offs can be made to enhance one aspect of system performance but, almost certainly, another aspect will be degraded. The two channel systems require the greatest bandwidth. However, they have been designed to maintain a higher level of compatibility and may have been subjected to compromises which do not use the bandwidth in the most efficient manner.

All of the proposed systems increase the vertical and horizontal resolution when compared with NTSC. However, all of them suffer in one way or another when motion is present in the picture. The CBS system would exhibit the least loss with motion. The Del Rey Group system would probably exhibit the greatest loss.

Three different techniques are used in the proposals to increase the vertical resolution. One system transmits two times the number of lines (CBS). Two of the systems transmit half of the information from two times the number of lines using an interleaved technique (MUSE, Del Rey Group). Three systems increase vertical resolution by improving the Kell Factor (Bell Labs, Glenn, Philips).

Three different techniques are used to increase the horizontal resolution. One system transmits two times the number of samples (Bell Labs). One system uses a greater bandwidth (CBS). Four systems transmit information from extra horizontal samples using an interleaved technique (MUSE, Glenn, Del Rey Group, Philips).

Five systems transmit more color information than is contained in the NTSC system (MUSE, Bell Labs, CBS, Glenn, Philips).

All of the proposed systems incorporate additional information to present a wide screen picture. All of the systems could use the 1125/60 studio signal as an input signal.

All of the proposed systems, except the CBS system, require a frame store in the receiver.

SUMMARY

In this paper we have looked at various proposals for delivering higher definition pictures to the public. The proposals range from improvements to the NTSC system to HDTV transmission systems requiring a greater bandwidth than is available in a single NTSC channel. While it seems quite likely that the 1125/60 HDTV studio system will become the studio standard, standards for delivery to the public are an open question.

The HDTV delivery system that has been most developed is being used by Japanese manufacturers to design consumer equipment. That system would most likely be usable for VHF or UHF terrestrial broadcasting. However, its use would raise a number of standards and regulatory issues.

Bandwidth considerations may lead to the use of different technical standards for different delivery systems.

Because of extensive misuse of the word "compatibility" I prefer to define a range of compatibility levels rather than define the word. I generally assume that higher levels of compatibility result in lower levels of performance for HDTV systems.

I made no direct comparisons between the proposed systems in this paper. Various system proponents make such comparisons in the reference documents for this paper; often in a competitive manner, stressing the benefits of the proponent's system and the weaknesses of the competitive systems. Care must be taken to understand the assumptions made in each case.

One final note. Dr. Glenn has compared the NTSC, improved NTSC, enhanced TV and HDTV systems in terms of achieving equal resolution on the viewer's retina. He assumes that improvements made to NTSC can also be made to enhanced and HDTV systems. His comparison is shown in Figure 25.

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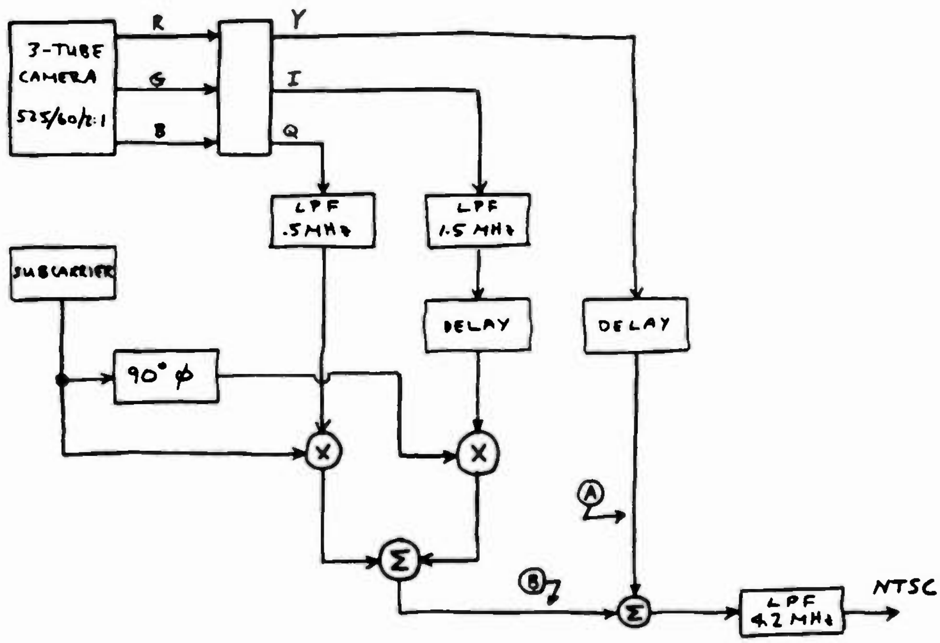
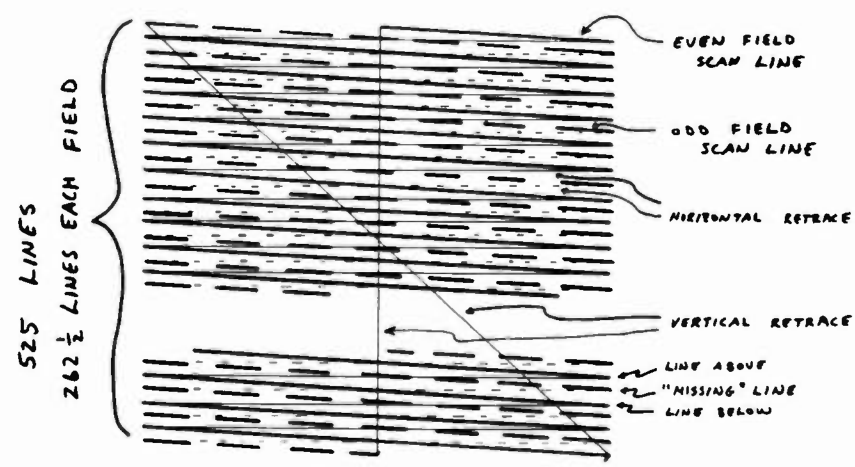


FIGURE 1 NTSC ENCODER



THE DASHED LINES ARE DISPLAYED IN 1/60 SECOND. THE SOLID LINES ARE DISPLAYED IN THE NEXT 1/60 SECOND. TO DISPLAY 525 LINES IN 1/60 SECOND, THREE POSSIBLE OPTIONS ARE:

A) DISPLAY EACH LINE IN THE FIELD TWO TIMES (POOR APPROXIMATION OF "MISSING" LINE)

B) DISPLAY THE AVERAGE OF LINE ABOVE AND LINE BELOW "MISSING" LINE (BETTER BUT NOT GOOD ENOUGH)

C) USE FIELD STORE AND HAVE PROPER "MISSING" LINE (VERY GOOD FOR STILL PICTURES BUT HAS MOTION PROBLEMS)

FIGURE 2 525 LINE 2:1 INTERLACE SCAN

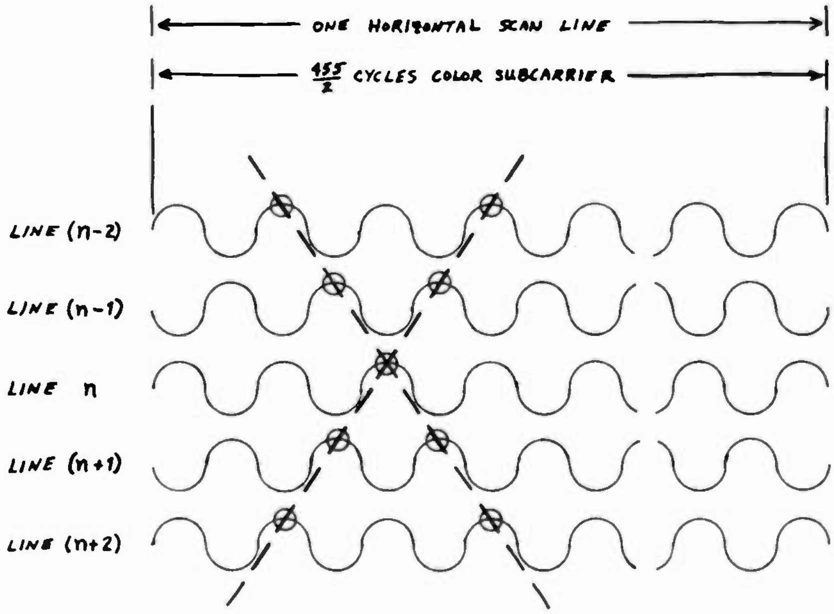


FIGURE 3 PHASE OF COLOR SUBCARRIER

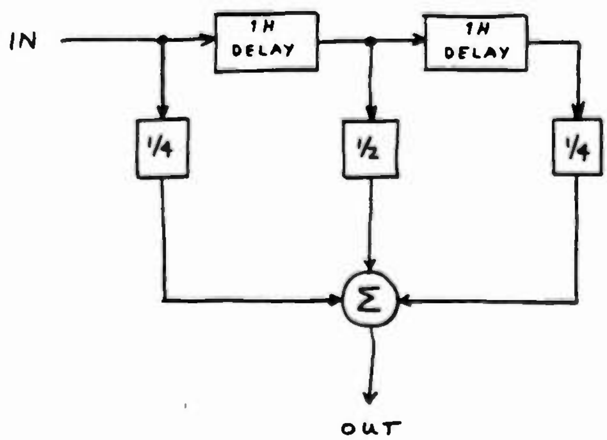


FIGURE 4 COMB FILTER

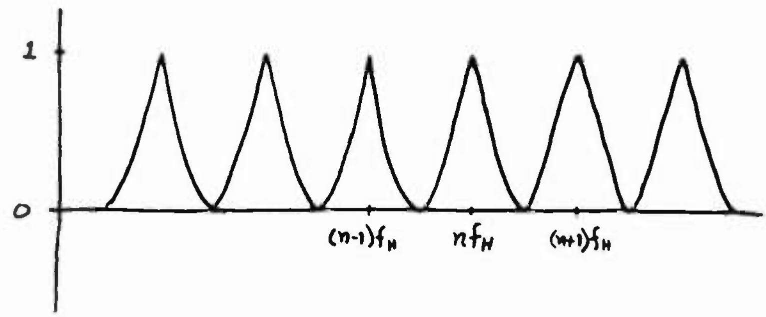


FIGURE 5 COMB FILTER OUTPUT SPECTRUM

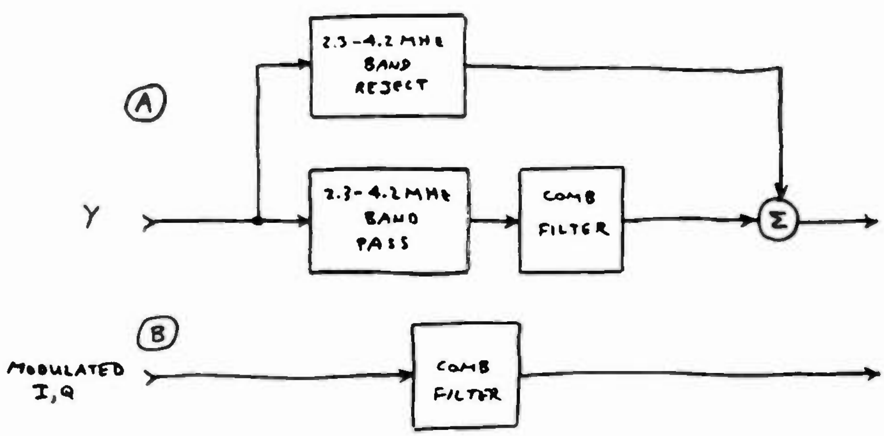


FIGURE 6 CIRCUITS TO "PRE-COMB" NTSC

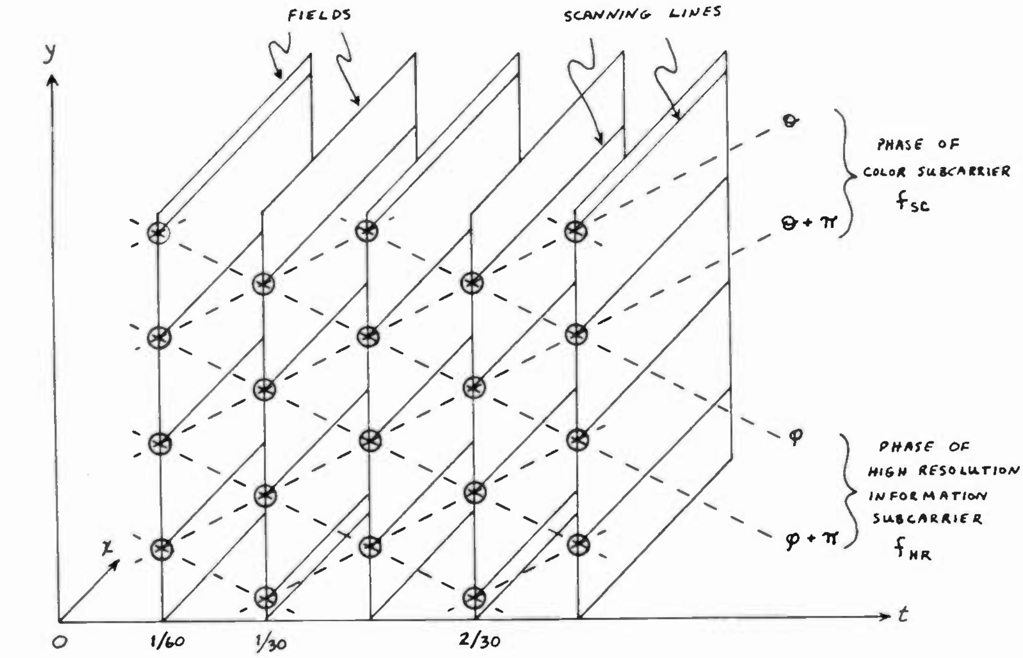


FIGURE 7   SCANNING LINE STRUCTURE, PHASE OF COLOR SUBCARRIER, PHASE OF HIGH RESOLUTION INFORMATION -- FUKINUKI PROPOSAL

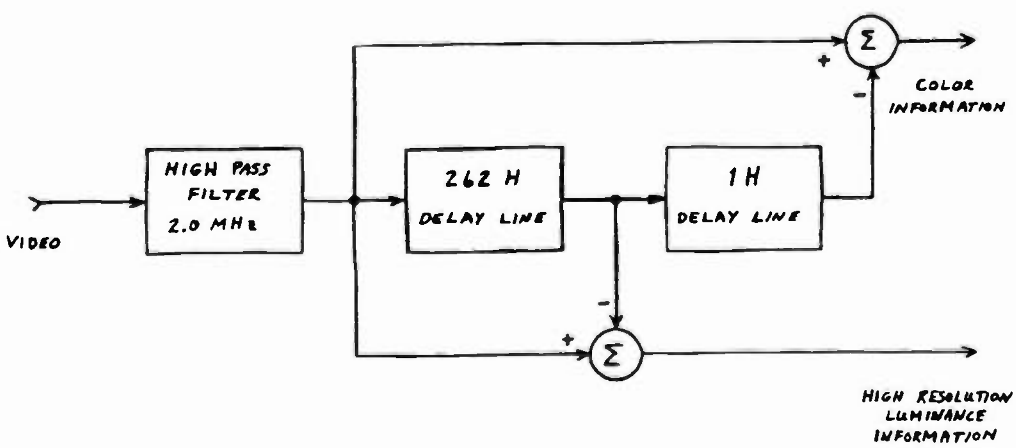
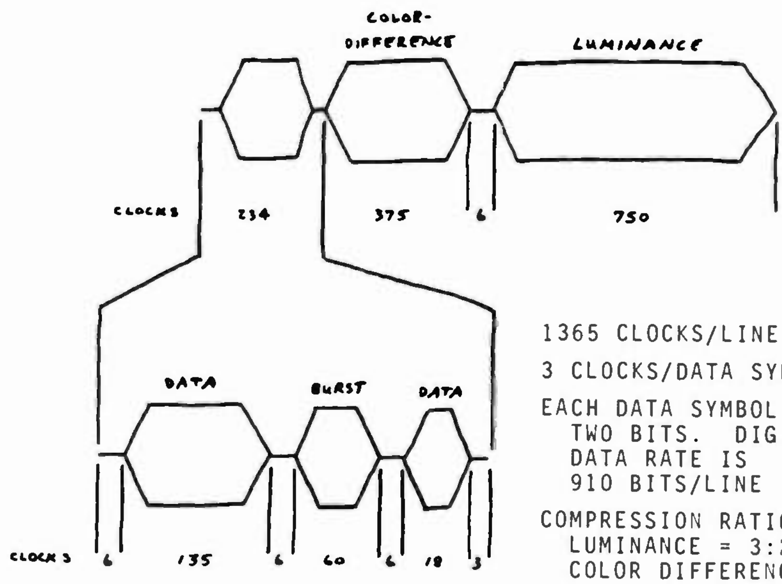


FIGURE 8   EXTRACTION OF COLOR INFORMATION AND HIGH RESOLUTION LUMINANCE INFORMATION -- FUKINUKI



1365 CLOCKS/LINE

3 CLOCKS/DATA SYMBOL

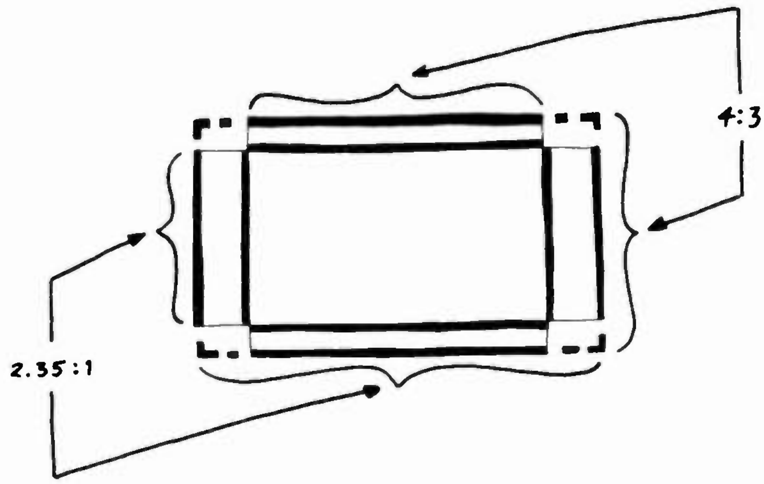
EACH DATA SYMBOL CARRIES
TWO BITS. DIGITAL
DATA RATE IS
910 BITS/LINE

COMPRESSION RATIO
LUMINANCE = 3:2
COLOR DIFFERENCE = 3:1

FIGURE 9 B-MAC HORIZONTAL LINE

1125	LINES PER FRAME
60	HERTZ FIELD FREQUENCY
2:1	INTERLACE
16:9	ASPECT RATIO
1035	ACTIVE LINES
1920	LUMINANCE SAMPLES / ACTIVE LINE
960	COLOR DIFFERENCE SAMPLES / ACTIVE LINE
?	HORIZONTAL BLANKING
?	SAMPLE FREQUENCY
?	COLORIMETRY

FIGURE 10 1125/60/2:1 STUDIO SYSTEM



4:3 & 2.35:1 RECTANGLES ARE EQUAL AREA

INNER RECTANGLE ASPECT RATIO = OUTER RECTANGLE ASPECT RATIO

$$= \sqrt{\frac{4}{3} \times \frac{2.35}{1}} = 1.770$$

NOTE THAT 16:9 = 1.777···

FIGURE 11 SHOOT AND PROTECT WITH 16:9 ASPECT RATIO

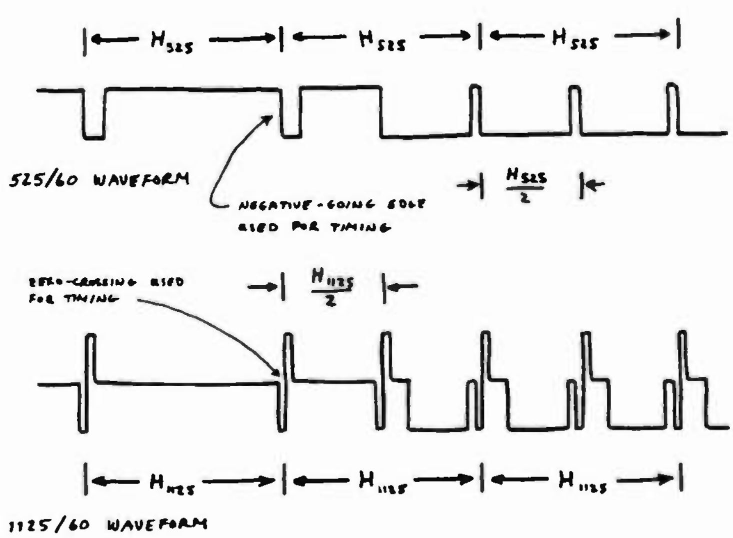


FIGURE 12 NHK PROPOSAL FOR 1125/60 SYNC WAVEFORM

INCREASING COMPATIBILITY ↑ / DECREASING COMPATIBILITY ↓

LEVEL 5	HDTV RECEIVED AND DISPLAYED AS HDTV ON NTSC RECEIVER
LEVEL 4	HDTV RECEIVED AND DISPLAYED AS HIGHEST QUALITY NTSC ON NTSC RECEIVER
LEVEL 3	HDTV RECEIVED AND DISPLAYED AS REDUCED QUALITY NTSC ON NTSC RECEIVER
LEVEL 2	HDTV RECEIVED WITH INEXPENSIVE ADAPTOR FOR DISPLAY ON NTSC RECEIVER
LEVEL 1	HDTV RECEIVED WITH EXPENSIVE ADAPTOR FOR DISPLAY ON NTSC RECEIVER
LEVEL 0	HDTV CANNOT BE RECEIVED FOR DISPLAY IN ANY FORM ON NTSC RECEIVER AND HDTV RECEIVER CANNOT DISPLAY NTSC

GENERAL RULE OF THUMB

HIGHER LEVELS OF COMPATIBILITY
PROBABLY RESULT IN LOWER PERFORMANCE
LEVELS FOR THE HDTV SYSTEM

FIGURE 13 COMPATIBILITY LEVELS

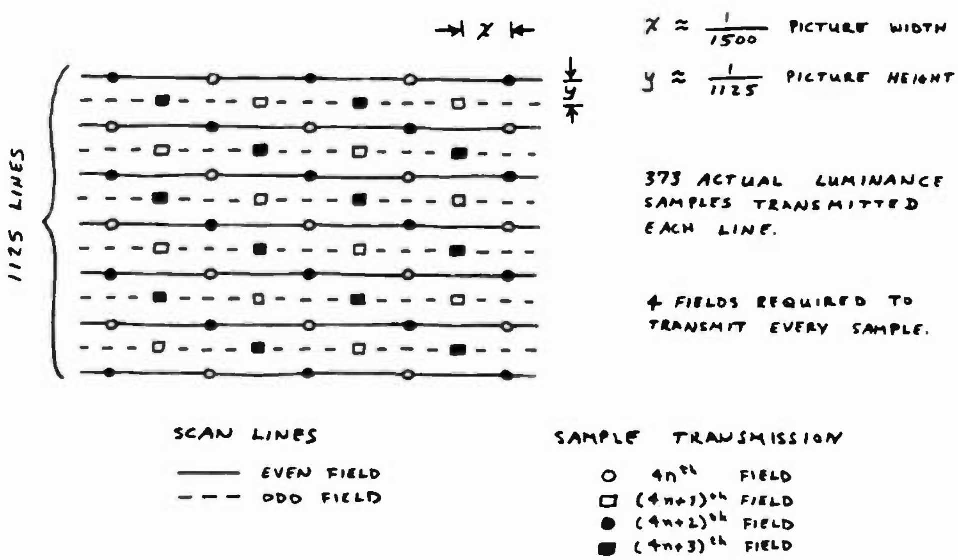


FIGURE 14 MUSE SAMPLING PATTERN

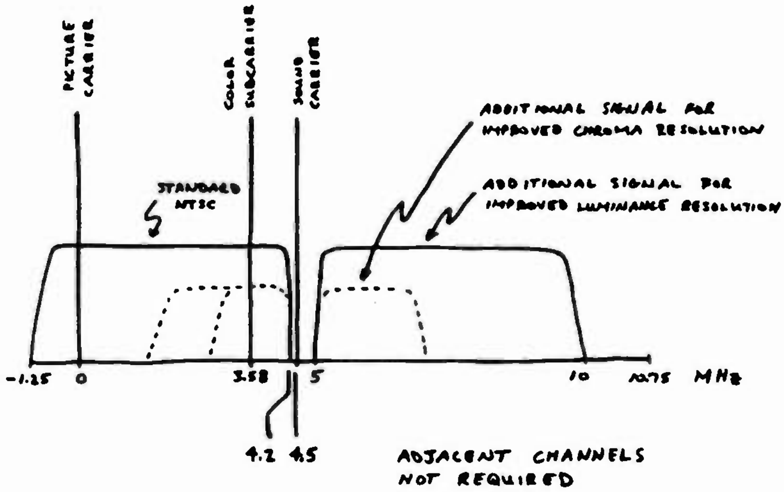


FIGURE 15 TRANSMITTED SPECTRUM -- BELL LABS

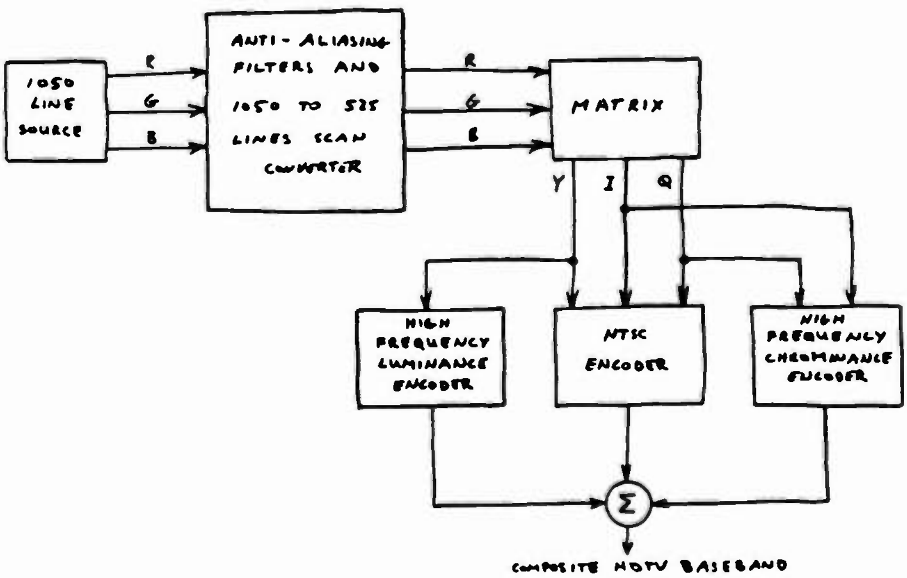


FIGURE 16 HDTV ENCODER -- BELL LABS

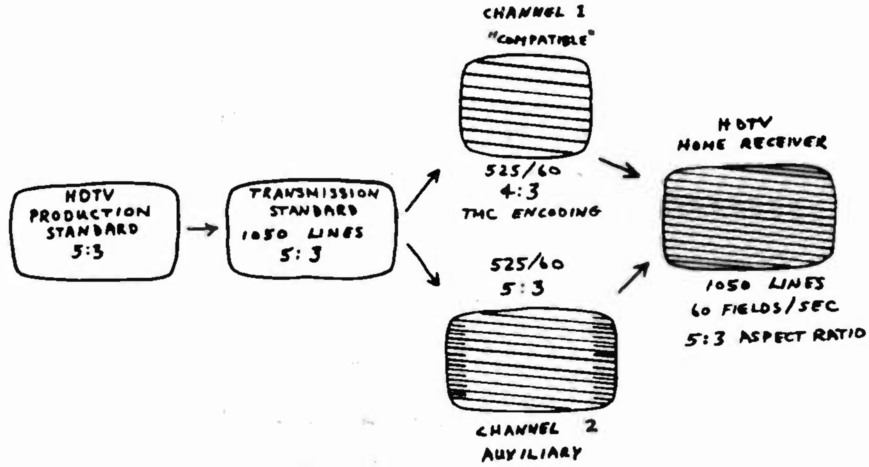


FIGURE 17 DBS HDTV 2-CHANNEL SYSTEM -- CBS

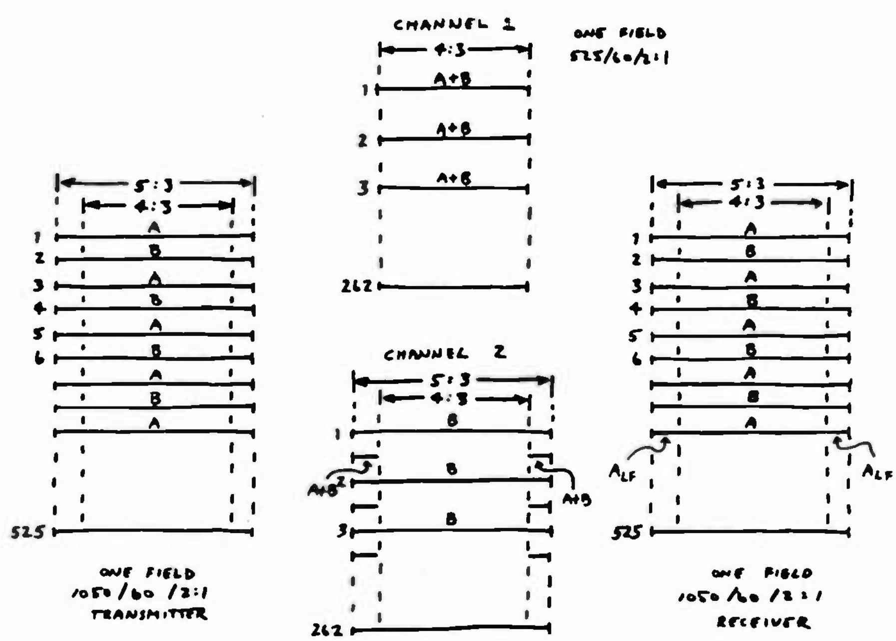


FIGURE 18 SIMPLIFIED SCANNING GEOMETRY -- CBS

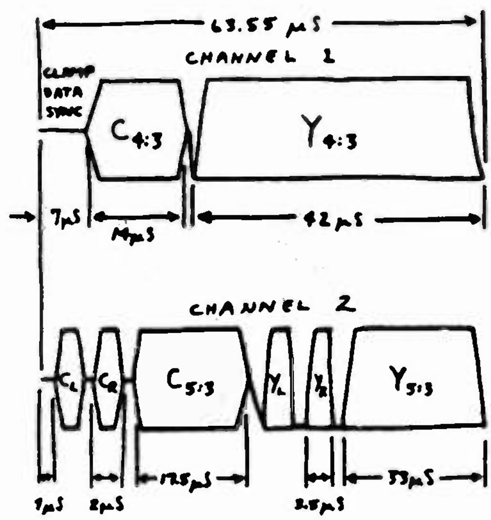


FIGURE 19 TMC FORMATS -- CBS

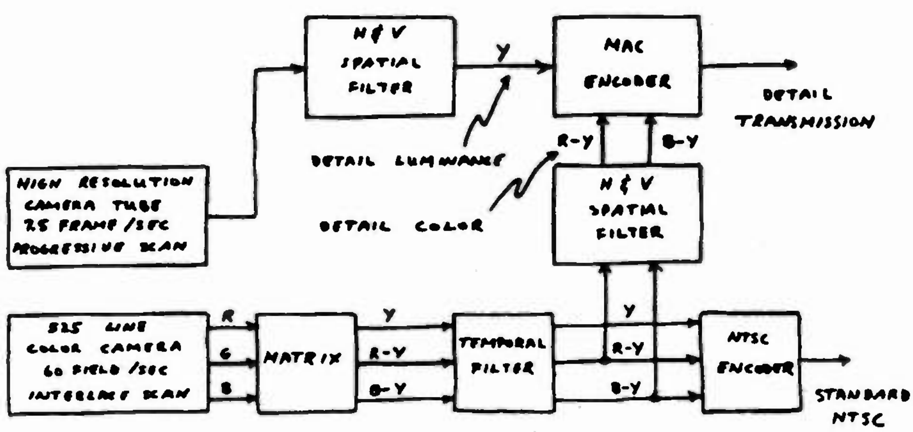


FIGURE 20 HDTV ENCODER -- GLENN

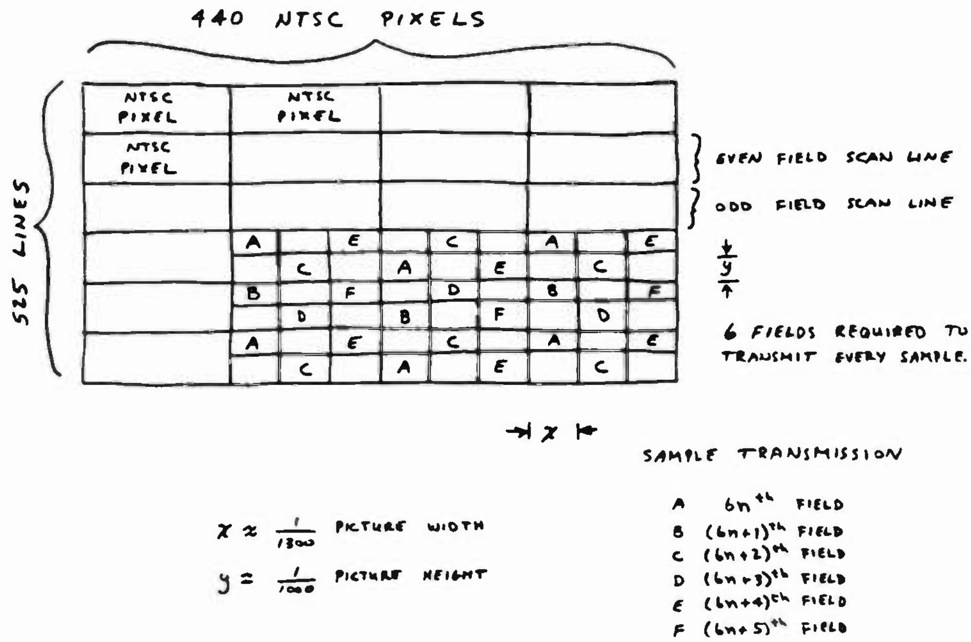


FIGURE 21 DEL REY GROUP SAMPLING PATTERN

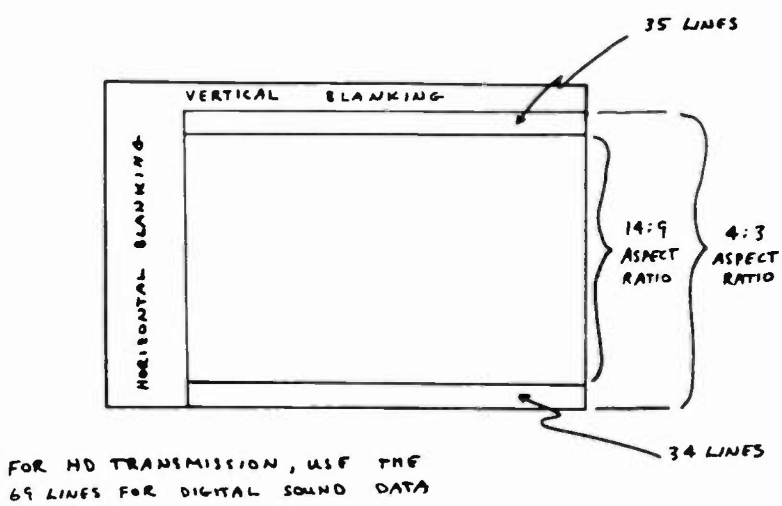


FIGURE 22 14:9 ASPECT RATIO -- DEL REY GROUP

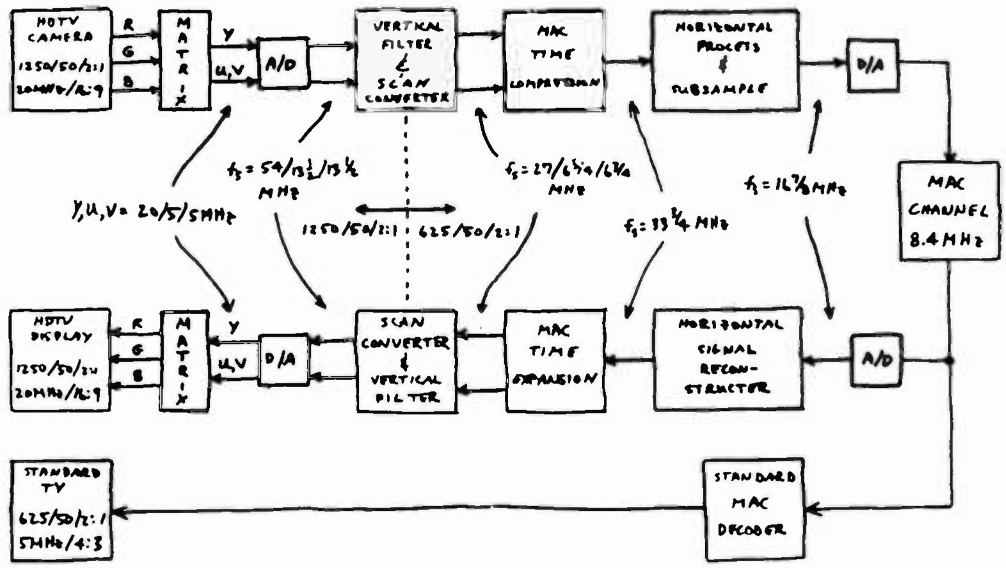


FIGURE 23 HD-MAC SYSTEM -- PHILIPS

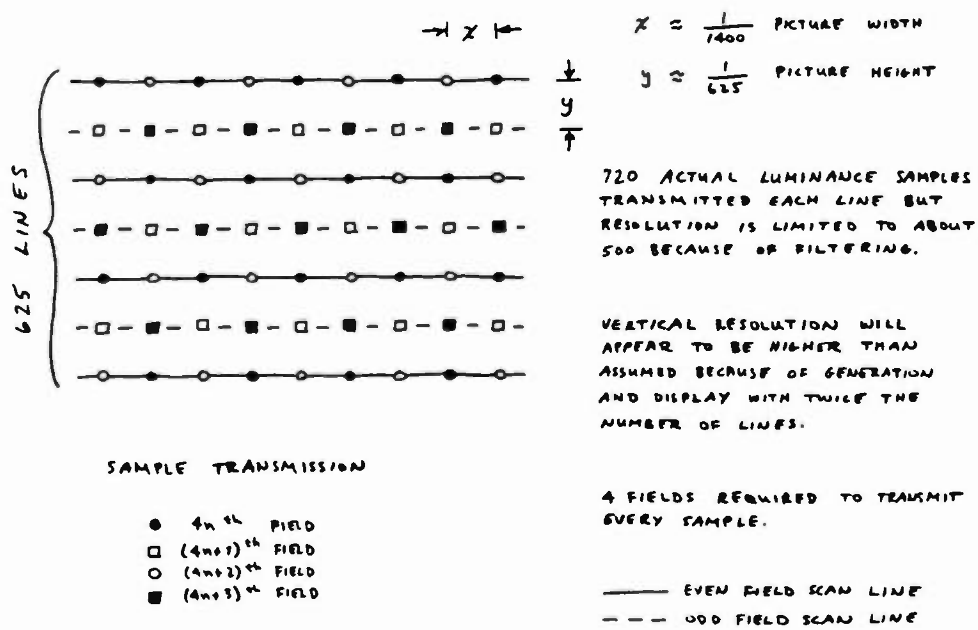


FIGURE 24 PHILIPS SAMPLING PATTERN

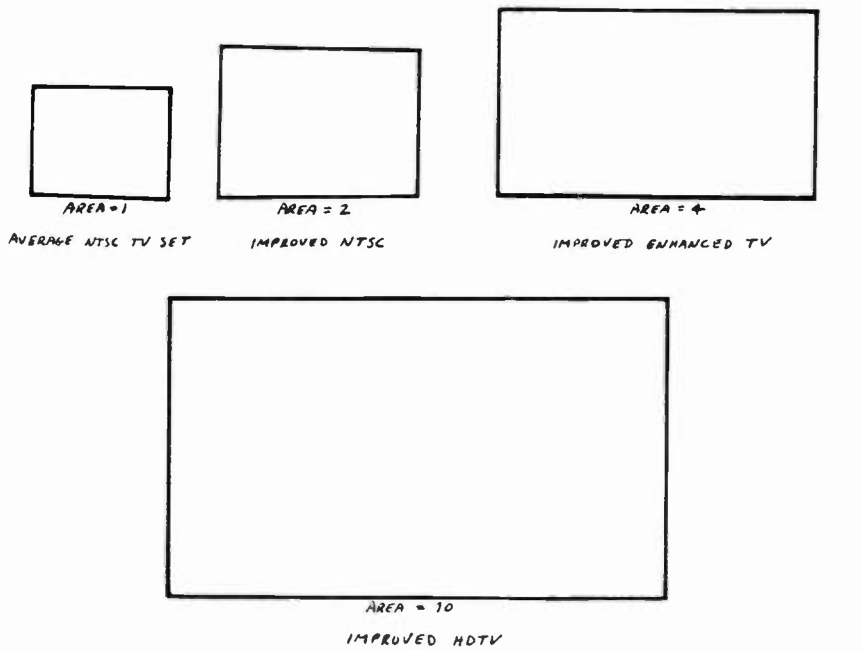


FIGURE 25 RELATIVE SIZES OF IMAGES DISPLAYED WITH SAME RESOLUTION ON VIEWER'S RETINA -- GLENN

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