1993 PROCEEDINGS

47th Annual
Broadcast Engineering
Conference Proceedings

National Association of
NAB
BROADCASTERS

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Broadcast Engineering
Conference Proceedings

Las Vegas, Nevada
April 18–22, 1993

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These proceedings contain technical papers presented at the NAB Broadcast Engineering Conference April 18–22, 1993.

Published by the NAB Office of Science and Technology

ISBN 0-89324-164-4

Additional copies of the 1993 Proceedings are available through NAB Services, 1771 N Street, N.W. Washington, DC 20036-2891. For price information, please write or call 1-(202)-429-5373.

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April 1993

Dear Industry Engineer:

On behalf of NAB's Engineering Conference and Advisory Committee, we are pleased to present the *1993 NAB Broadcast Engineering Conference Proceedings*.

NAB's 47th Broadcast Engineering Conference features useful and informative presentations to help you cope with the challenges facing our industry. The conference focuses on the practical applications of existing technologies, the new opportunities offered by emerging technologies, such as data broadcasting and interactive television, and professional development for engineering managers.

Data broadcasting technologies offer new opportunities for both television and radio. In addition to the Radio Broadcast Data System (RBDS), technologies such as CouponRadio, NHK's high capacity FM subcarrier, Interactive Video Data Service (IVDS), and TV Data Systems (TVDS) present real potential for increased station revenue.

The management training element of this conference is particularly important as engineers continue to face the challenges of developing long range strategic plans while running a station on an ever-shrinking budget. To be successful they must maintain a sharp technical edge without losing a clear vision of business realities.

We again welcome the participation of the Society of Broadcast Engineers in programming five sessions for the conference. This year also marks the first participation of the Institute of Electrical and Electronics Engineers' (IEEE) Broadcast Technology Society which has created two sessions discussing the development of broadcast standards. We also welcome the continuing strong participation of the international broadcast engineering community providing a global perspective on the broadcast industry.

Please share your comments and suggestions for our conference. Feel free to call or write at any time.

Best regards,

Michael Rau

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Senior Vice President
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Sunday, April 18, 1993

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DIGITAL AUDIO SYSTEMS

Sunday, April 18, 1993

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*Paper not available at the time of publication.

DIGITAL AUDIO WORKSTATIONS

Sunday, April 18, 1993

Moderator:
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New York, New York

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Doug Simpson
Crouse-Kimzey
Fort Worth, Texas

*Paper not available at the time of publication.

DIGITAL AUDIO WORKSTATION NETWORK—A REPLACEMENT FOR AUDIO CARTS AT CBS TELEVISION NETWORK

Gregory M. Coppa
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New York, New York

***Abstract* - CBS has implemented an automated Digital Audio Workstation Network (DAWN) cart system for record and playback of audio announcements within their Broadcast Origination Center (BOC). DAWN, which replaced antiquated NAB carts, provides CBS with a high quality reliable audio cart system that has streamlined audio cart management.**

1. INTRODUCTION

In June of 1991 CBS began distributing its Network television signals from a fully automated Broadcast Origination Center (BOC). BOC provided CBS with a program and commercial integration facility that allowed for origination of up to ten networks, all under computer control.

The BOC design included state of the art equipment: multicassette D2 library management systems, networked video still stores and Local Area Network (LAN) based machine control. Yet, at the heart of the BOC audio system was an old workhorse -- the NAB cart, complete with its twenty-four volt control interface, Cinch Jones type connector and characteristic wow and flutter.

The time was right for replacing this workhorse but was the technology? Not during the initial design phases of the BOC. At NAB 1991 CBS saw a demonstration of a Gentner Communications product -- DAWN: Digital Audio Workstation Network, that seemed to meet some basic BOC design requirements -- it was PC based, easily automated, would accept a playlist downloaded from automation, play carts from the list when commanded and finally it potentially could streamline cart management thereby reducing costs. Thus, CBS decided to use the DAWN product to replace the NAB carts.

A description of the CBS DAWN system follows. Section 2 discusses the DAWN hardware, section 3 details the audio workstation software and examines the operation of DAWN at CBS.

2. DAWN HARDWARE

DAWN is a local area network of audio workstations. The network consists of a file server, audio workstations, a remote access server, automation interface computers and a programmable PC keyboard. A block diagram of the LAN showing a typical workstation interface appears in Figure 1.

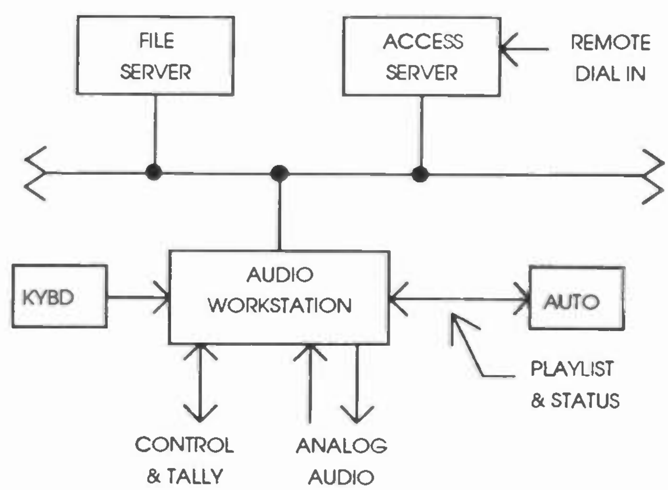

Figure 1. DAWN system block diagram.

The CBS DAWN system has eight audio workstations. Six of these supply the ten BOC channels and are under automation control. A workstation dedicated to the BOC announce booth is a cart record location. The last workstation, located at CBS Hollywood, is a remote record location.

A. File Server.

The file server is a 386 based AT compatible computer running Novell's Netware 386 network operating system. Netware supports simultaneous file sharing, shared printing facilities, system performance monitoring and is responsible for network security.

The server provides central storage for all digitized audio cart files and related text data. This has several advantages -- it gives each workstation access to the most recent version of a cart, streamlines cart management and has made data backups easier to manage.

B. Audio Workstation.

The audio workstation is also a 386 based AT compatible. It contains the standard peripherals: VGA, serial and parallel ports, and a hard disk. Installed in the PC are two additional pieces of hardware. One is a network interface adapter that supports LAN communications and the other is a Digigram PCX3 Digital Audio System board.

The PCX3 is a real time audio compression/decompression system that uses either the MUSICAM or WB48 SBC algorithm. It is capable of stereo audio processing at a sampling rate up to 48 khz with 16 bit resolution. Several compressed data rates are available. 128 kbits/sec per mono channel is typical and corresponds to a compression ratio of 6 to 1. The CBS DAWN system uses the WB48 algorithm with 48/16 sampling and 6 to 1 compression. For a 30 second stereo audio spot this leads to MS DOS file sizes on the order of one megabyte.

BOC automation interfaces to the workstation via the parallel port of the PC. A relay contact closure from automation commands the workstation to play a cued cart. Cued and play tally are returned, again via contact closure, to automation which displays this information to a BOC operator.

C. Programmable Keyboard.

Each automated workstation shares two keyboards, Figure 2. The local keyboard is a standard PC keypad. The remote keyboard, located in a BOC control room, is software configurable.

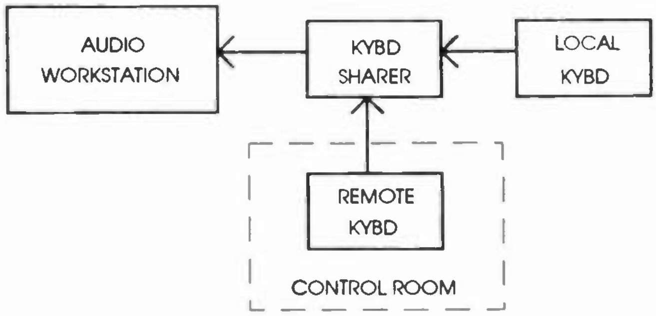

Figure 2. Workstation keyboard architecture.

The control room keyboard does not require all DAWN functions, therefore only those necessary for on air use are programmed. DAWN functions requiring several keystrokes are programmed as macros and executed using a single keystroke. This greatly simplifies the operational interface and reduces the potential for errors.

D. Automation Computer.

Attached to the RS-232 communications port of the workstation is the automation computer. This computer, also a PC compatible, communicates with the BOC automation system reporting workstation status and downloading playlists.

E. Remote Access Server.

Remote access to the DAWN system is provided through an access server which supports eight dial up phone lines. Remote users have modems and software on their local PC's which support the remote session.

3. DAWN SOFTWARE AND OPERATION

The DAWN software, Point & Shoot, is a custom application written by Mediacomp of Ontario Canada. The software includes cart management functions, interfaces to the PCX3 board, providing cart record, playback and editing facilities and interacts with BOC automation.

A. Cart Management

Point and Shoot streamlines cart management procedures by consolidating cart information and by providing managers easy access to this information. Cart management functions include searches, printouts of billing and announcer royalty reports, and automatic purges of stale carts.

All cart data is centralized on a workstation into a cartwall. The cartwall is a collection of 4000 bins. Each bin houses the information of an individual cart. A bin is made up of the following fields: title, shelf life, announcer, and cart type. The title is a description of the cart, the shelf life specifies the starting and ending dates a cart is to be active, the announcer field identifies who made the recording and finally the type field is used for internal billing information. A bin also contains storage and editing facilities for scripts.

A search function permits rapid location of an individual cart. Searches are conducted on the bin number, title, date or announcer fields. The search function will find all occurrences of a given string regardless of which field it is in.

Each cart has a shelf life which denotes the dates a cart is active and can be used for air. Every day a purge of the DAWN database is conducted automatically using the shelf life date to determine if the cart should be removed. The auto purge process keeps the database from becoming filled with stale data and does not require a manager to continually maintain the database.

The cart creation process at CBS consists of several steps. First an order must be placed by the CBS Hollywood production department for a cart. The order generates the bin number, title, shelf life and type of cart. Upon receipt of the order, a media manager dials into the BOC DAWN system using a desktop PC and software that allows him to control a DAWN workstation in New York. The cart information is entered into the cartwall and then transferred to the file server. The manager disconnects from the system and sends the order to the creative department.

The creative department writes the script for the cart using their favorite PC word processor. When the script is complete the writer dials into the BOC DAWN system and downloads the text to the appropriate bin.

The cart is now ready to either be recorded or read directly to air. If it is to be recorded, an announcer gets a work order which lists the bin, enters his initials and records the cart. The script may also be read directly to air depending on schedule and time requirements. Downloading the script has eliminated the need to deliver a hard copy to the announcer, streamlining the cart process.

Other mangers at CBS use information stored on the DAWN system for payment of announcer royalty and to track billing. They also have access to the system via modem and can browse the cartwall. Reports can be generated which detail such information as what royalties an announcer is due, or what department should be billed for a cart. This has led to more accurate tracking of the costs and revenues associated with cart production and broadcasting.

B. Cart record, playback and editing.

Recording a cart on the workstation is similar to recording on an analog tape recorder. The record process begins either manually, with the press of a button or automatically when the audio level has risen above a user selectable value. As the recording proceeds, the cart duration and a bar graph of the right and left channels are displayed on the workstation screen.

Once recording is complete, random access allows the user to cue to any location within the spot. Cueing is facilitated using typical tape machine operations such as rewind and fast forward. Also, a digital scrub function, with variable speeds, allows the user to jog through the recorded cart for easy cueing.

A cart's duration can be trimmed using virtual in and out points. A virtual point is created by scrolling to the desired location and simply marking the spot. No further recording is required. The software recognizes the virtual points as the place where playback should begin and end. Virtual marking speeds the process of cart creation by reducing the number of retakes necessary to meet schedule duration requirements. Durations can also be modified using a process known as squashing. Squashing removes segments of audio from the recording which are below a user selectable level.

All cart recordings are initially stored on the workstation's local hard drive. Edits to a cart are made using the local copy. When the editing process is complete the data is automatically transferred to the file server making the edited version available to all workstations.

At CBS, cart recordings originate from several locations, the BOC announce booth, announce booths located within the New York plant, and CBS Hollywood. The BOC announce booth has a dedicated workstation which an announcer uses to record and edit a cart. The other New York locations are connected to the BOC system using analog audio ties. A BOC operator, in communication with the remote announcer, records and edits announcements made from these sites.

The CBS Hollywood location has a stand alone DAWN workstation that records carts and then downloads them via high speed modem to the BOC file server. The modem communicates at 19.2 baud over a dial up phone line with a typical data transfer rate of approximately 100 kbytes per minute. For a 30 second stereo cart this translates into a transfer time of about ten minutes. Loading carts in this manner has reduced shipping and handling costs, and eliminated shipping delays, improving quality and accuracy of on air announcements.

C. BOC Automation.

On air cart playback from a DAWN workstation can occur either manually or from automation. Manual or automated control is selected by a BOC operator using the remote keyboard.

Automation generates a playlist of carts to be aired from the BOC channel schedule by looking into the schedule to determine which events are cart events. It builds a list of the carts, which have been specified in the schedule by bin number, based on the time the cart is to air. The list is then transmitted to the appropriate DAWN automation computer which sends the playlist on to the workstation.

When Point and Shoot receives a playlist from automation it builds the list using carts stored on the file server. It looks for a cart by its bin number and if the cart exists the cart data is copied from the server to the local hard drive. If a cart is missing a place is created for it in the playlist and a missing cart status is sent back to the BOC operator. Once the playlist has been built Point and Shoot now waits for a command from automation to play the cart at the top of the list. Meanwhile, the automation computer is continually monitoring the workstation, reporting its status to a BOC channel operator.

Carts are played when a play command is detected from automation. This command interrupts the workstation and plays the cued cart. During playback, tally status is reported to automation and is displayed on the channel schedule. Once a cart has been played it will be erased from the local drive provided it does not appear in the playlist at a later time. The next cart to air will then cue and wait for the next play pulse from automation.

Interruption of automation control may sometimes be required. This is particularly true when special news events occur. When this happens, the channel schedule is frozen at the last event and an announcement is played indicating that a special report is about to air. In order to facilitate this process a special "jingle playlist" has been created. A jingle playlist contains several "canned" announcements that introduce the sand end the special report. These are played manually under operator control. When the special situation has ended the jingle playlist is completed and control is returned to automation.

D. DAWN Flexibility.

As mentioned above a design requirement of the BOC was that devices should be PC based. This is due to the fact that a PC system is inherently flexible. A great deal of support and off the shelf software exists which improves the productivity, maintainability and reliability of the system. The DAWN system has proven to be quite flexible in ways CBS had not initially intended. For one, the demand for remote access by managers, which has led to the development of more efficient cost accounting and reduced cart handling costs, was never anticipated. Off the shelf software and simple modem interfaces made this possible with little or no added cost.

New procedures continue to be developed. A proposal for checking the channel schedule well in advance of its air time is now under consideration. This process would check the DAWN cart inventory to determine what carts are not available. A list of these carts will be generated alerting management as to which carts need to be recorded. The development of this type of process is expected to cost very little because the cart database is readily available and easy to access. Yet, its impact in reducing lost revenue will improve the bottom line.

5. CONCLUSION

NAB carts have been replaced at the CBS Television Network using an automated digital audio workstation network. The performance of these audio workstations has improved the quality and reliability of announcements.

Remote access to the DAWN system has streamlined cart management procedures and reduced the costs associated with cart production. Easy access to cart information has improved billing accuracy and royalty accounting. Workstation automation has improved the reliability and accuracy of on air cart playback.

The flexibility of a PC based system is providing CBS with new opportunities for using the data stored in DAWN. An inventory reconciliation program is being developed which will provide CBS management with timely schedule information reducing lost revenue costs due to missing carts.

The DAWN system is now the cart workhorse for CBS. It is expected that it will serve as well as the NAB cart and for as long.

References:

1. DAWN Product Training Manual.

DIGITAL AUDIO PROCESSING

Sunday, April 18, 1993

Moderator:
Carl W. Davis, Voyager Communications, Inc. Raleigh, North Carolina

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THE LEAST TREATMENT PRINCIPLE AND ITS APPLICATION TO SINGLE-ENDED SIGNAL PROCESSING
Stan Cossette
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THE LEAST TREATMENT PRINCIPLE AND ITS APPLICATION TO SINGLE-ENDED SIGNAL PROCESSING

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ABSTRACT

A design principle used in the development of professional-quality, complementary noise reduction systems is described. The benefits of applying this principle to single-ended signal processing tasks such as compression and expansion is presented. Comparison to simpler design methods is also presented along with several figures showing advantages of applying this principle.

I. INTRODUCTION

In his paper describing the Spectral Recording process, Ray Dolby coined a term for an underlying principle he used to guide his design[1]. He was faced with the task of designing a professional quality, complementary noise reduction (NR) system that would surpass all existing systems including his own. Considering the encoder, he reasoned that the best design would be one which could boost the input signal by a fixed amount under the greatest number of signal conditions. If required to reduce the boost to prevent channel overload, the system should not do so at other frequencies away from the dominant signal. Furthermore, the system should reduce the signal boost only by the amount necessary. The principle was dubbed the Principle of Least Treatment. It is no accident that the well-received Spectral Recording process adheres more closely to this principle than any previous NR system. This principle was later applied in the design of the S-type NR system for consumer recording formats[2].

While this principle was used to develop a complementary NR system, it has proven to be a useful philosophy for developing other signal processors as well. In this paper, I will discuss a couple of common signal processing examples—showing the pitfalls of common designs and the benefits of applying the Least Treatment Principle.

II. BENEFITS OF LEAST TREATMENT APPLIED TO COMPRESSION

A common task required in audio signal processing is the task of dynamic range compression. Here the requirement is to reduce the dynamic range of the input signal by some amount for further transmission, recording, or as an artistic effect. Typically, the user is given some control over where the compression begins (threshold) and the compression ratio encountered above the threshold. Often control is also provided over the attack and decay time of the processor. The latter two controls are usually provided as a way of minimizing the negative audible effects that are produced by the process. The steady-state response of the compressor can be characterized by plotting its output versus input. This is typically called a transfer curve and is shown in Figure 1.

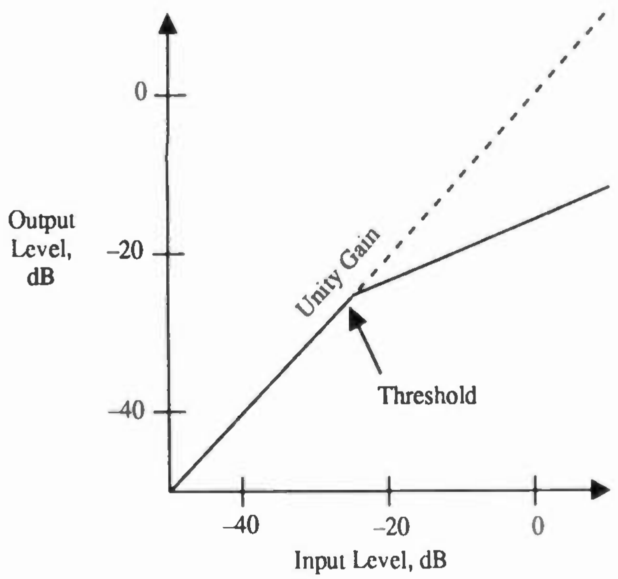

Fig. 1 Simple Compressor Transfer Curve

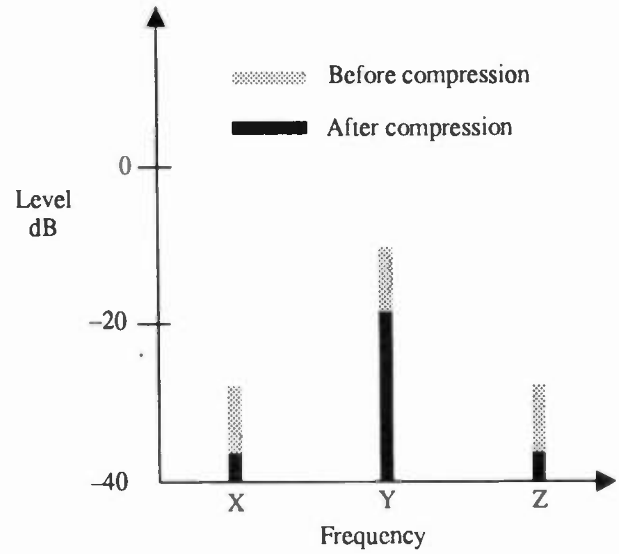

Fig. 2 Simple Compressor Response to Complex Tone

The pitfalls associated with this process are "pumping" and distortion. Pumping is the common term for the case of a dominant signal causing amplitude-modulation of a smaller signal. The modulation follows the envelope of the larger signal and causes the level of the smaller signal to "pump" up and down. Distortion typically appears as modulation distortion which can occur on certain steady-state signals or during transients.

Pumping is caused by using a compression process that is not able to compress by different amounts at different frequencies. If a simple, wide-band compressor is used, all signal components are attenuated by the amount required by the dominant signal. In Figure 2 this is illustrated for a complex signal made up of components X, Y, and Z. The figure shows that the 10dB attenuation required by component Y is imposed on the two smaller components. If components X and Z existed alone, they would not experience any compression.

Distortion of steady-state signals is usually caused by allowing the compressor control signal to change too rapidly. Inasmuch as the compressor stage is an amplitude modulator, if the modulation frequency is too high, the sidebands of the modulation will become audible. A good compressor must be able to respond to the signal envelope while not reacting to the signal itself. This is difficult since audio signals span a thousand to one frequency range.

Excessive attenuation due to compression can cause problems in two ways. First, the basic laws of compression require that when the input signal changes in level the compression effect (attenuation) must change to that required by the new signal. If the attenuation change takes place too slowly, there will be large overshoots. If this change takes place too quickly, there is a risk of modulation distortion. If there is no limit on the amount of gain-change that can take place, it will be nearly impossible to make the change inaudibly. Secondly, the more attenuation imposed by the compressor, the greater the chance of audible side effects such as pumping. Limiting the maximum amount of attenuation, hence level change, reduces the severity of these problems.

By applying the Least Treatment Principle (LTP) to the simple compressor, a few improvements can be made. First, the transfer characteristic can be changed to allow an upper limit on the amount of compression. Second, by using band-splitting or other techniques, the compression effect can be more frequency-selective, allowing attenuation in one band while none in another. Bandsplitting also allows changing the control-path reaction time in each band to correspond to the signals within that band. (Note: In some cases, an upper limit cannot be set due to other requirements.)

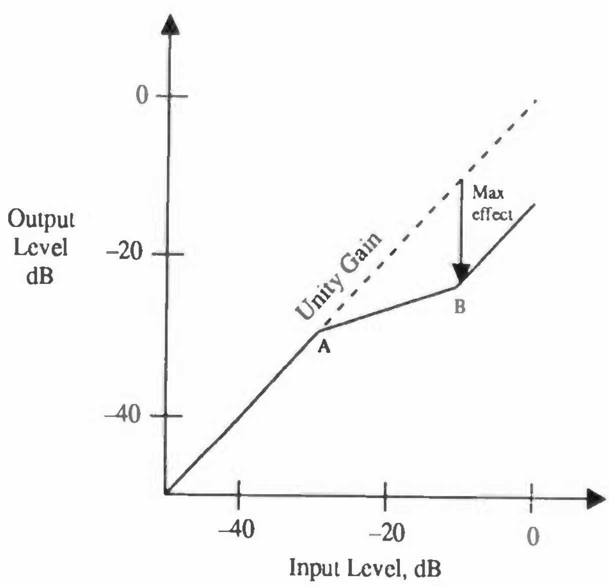

Fig. 3 LTP Compressor Transfer Curve

Figure 3 illustrates how the compression effect could be limited. Note that once point B is exceeded, no additional compression will take place. This can have a beneficial effect by reducing the need for fast attack and recovery times as well as minimizing other side effects caused by excessive attenuation. In Figure 4, the complex signal is compressed again but this time only the dominant signal component Y is attenuated by the required 10dB while the others are left "untreated". Application of the Least Treatment Principle during compression leads to fewer audible artifacts resulting in a compressed signal that does not "sound" compressed. This in turn allows creative use of compression effects in studio production or post-production for enhancing harmonics, dynamic equalization, intelligibility improvement and many other effects. Essentially, new applications are made possible with this type of processing.

A product has been designed using the Least Treatment Principle to enhance the low-level details in a recording such as the harmonics in an acoustic guitar or to enhance the intelligibility of a voice. Called the Dolby Spectral Processor, it is capable of boosting low-level signals while leaving high-level signals relatively unprocessed. A conventional compressor would behave as shown in Figure 2. It would not be capable of boosting low-level signals while a high-level signal was present. Using the Least Treatment Principle, the Spectral Processor is able to reduce the boost applied to the high-level signal (Y) while leaving the low-level details (X and Z) fully boosted as shown in Figure 4.

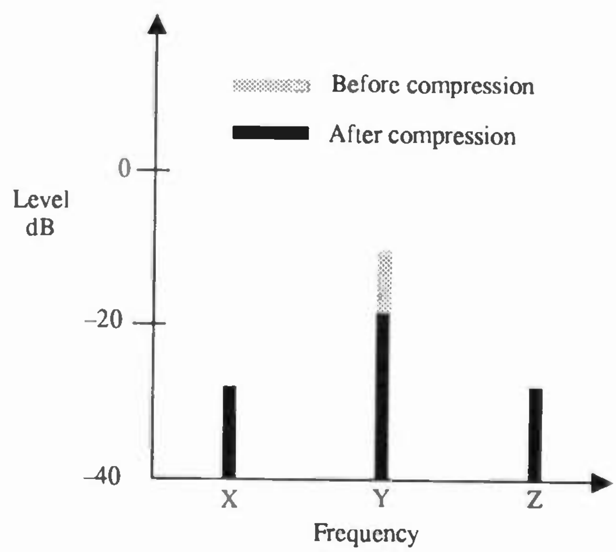

Fig. 4 LTP Compressor Response to Complex Tone

III. BENEFITS OF LEAST TREATMENT TO SINGLE-ENDED NR (EXPANSION)

Complementary noise reduction systems designed for transmission or recording have the advantage of processing the signal before it is corrupted by noise or distortion. Unfortunately, there are many cases where noise or background sounds are already present in a recording so there is a need for single-ended NR systems. These products have the difficult task of removing some unwanted signal such as noise or hum while retaining the desired signal. To make things even more difficult, the desired signal ideally is not audibly changed by the NR process itself.

Products that address this problem often use an expander whose threshold and expansion ratio are adjustable. In order to avoid unwanted side-effects the user is given control over attack time, decay time or sometimes frequency-weighting of the control path.

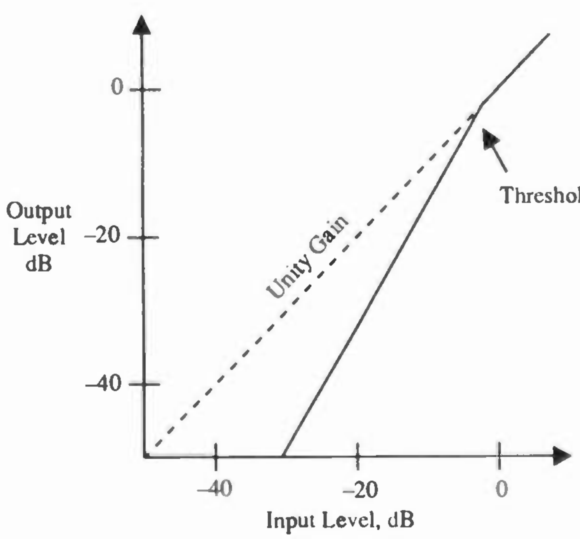

Fig. 5 Simple Expander Transfer Curve

A simple expander transfer curve is shown in Figure 5. Note that the expansion characteristic begins once the signal falls below threshold. Note also that there is no limit on the amount of expansion possible for very low-level signals. The process is commonly called 'downward expansion' because the expansion characteristic operates only to reduce rather than increase the signal level.

If a single expansion band is used, the dominant signal must control the attenuation amount as in the case of the simple compressor. This again leads to modulation of low-level signals by dominant signals. This effect is illustrated in Figure 6. This figure shows the gain change experienced by a noise signal while the wanted signal changes in level over a 15dB range. Note that the desired signal now changes by 30dB while the noise signal-level changes by about 15dB. While the noise may be made inaudible in the absence of the desired signal, it will certainly reveal itself when the desired signal is present.

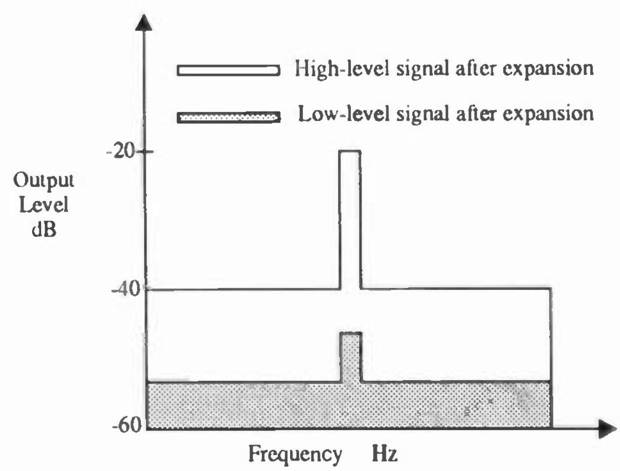

Fig. 6 Simple Expander Response to Tone with Noise

By limiting the amount of expansion and making the process more frequency-independent, better adherence to the goal of least treatment is achieved. Figure 7 shows the effect on a noise signal in the presence of a changing, dominant signal like that of Figure 6. First, the low-level signal is not allowed to drop indefinitely but is held at a fixed attenuation as determined by the expansion transfer curve. Ideally, the noise is reduced to an inaudible level. Reduction beyond this amount is unnecessary and can lead to problems. Then, in the presence of a high-level dominant signal, the noise is held at this low level. In this way, a constant, low-level noise floor is achieved even in the presence of large dominant signals.

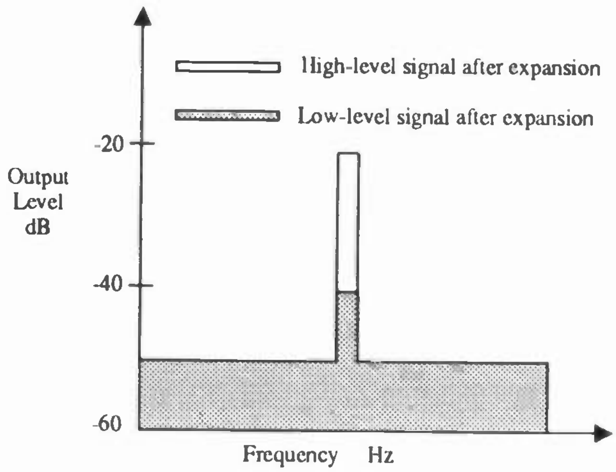

Fig. 7 LTP Expander Response to Tone with Noise

The Dolby Model 430 Background Noise Suppressor, designed primarily to reduce the unwanted acoustic noise often present in location recordings (traffic rumble in a period drama for example) was also designed using the previously mentioned goals. Here, the Least Treatment Principle allows the Model 430 to reduce background noise by a subjectively constant amount, even in the presence of speech. Conventional expanders, while reducing the noise well below inaudibility during pauses, would allow it to become audible in the presence of speech. Such a modulated background noise is often more intrusive than the original constant background noise.

IV. HELP FROM MASKING

Building a processor that perfectly embodies the Least Treatment Principle would be very difficult if it were not for some help from a psychoacoustic phenomenon known as masking. This phenomenon works in such a way that low-level signals become inaudible if they are near a much louder signal in frequency. The low-level signals are said to be 'masked' by the high level signal.

Design of a perfect compressor or expander would require that signals that are arbitrarily close in frequency are able to receive completely different amounts of compression or expansion. In order to build such a frequency-selective circuit, very high-order filters would be required. These filters would be complex and expensive and would require a separate compressor stage for each band. Building such a circuit would be very costly and may lead to problems with phase shift.

Masking relaxes the need for extreme frequency-selectivity in both compression and expansion. During compression, masking will hide level changes in the low-level signals that occur because the process may not be quite as frequency-independent as needed. During expansion, low-level signals are typically the target unwanted signal and masking will hide them in the presence of dominant signals. Masking essentially allows the design of a useful processor without the use of high-order filters and their accompanying phase-shift, complexity, and cost.

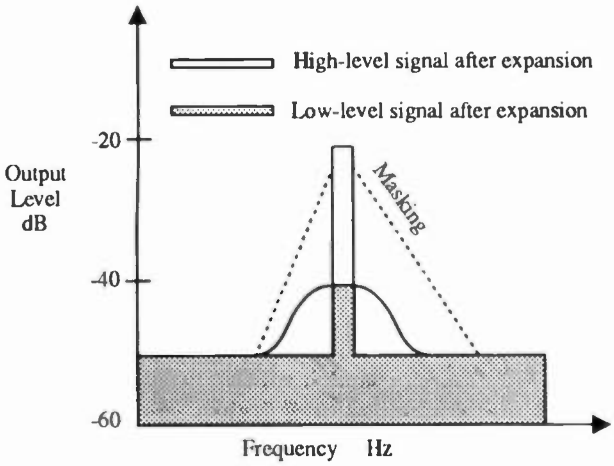

Fig. 8 LTP Expander Response
Showing Masking Effects

Figure 8 illustrates a more realistic expander response to the signal conditions of Figure 7. It can be seen that the noise floor does indeed rise in the presence of the high-level signal. Because the processor is frequency-selective however, this rise only occurs near the frequency of the high-level signal. Figure 8 shows a typical masking curve for a mid-frequency, high-level signal. The signals that fall underneath this curve would be masked. In this ideal example, the noise change is completely masked by the high-level tone. In practice, this ideal can be approached quite well over a wide variety of signal conditions without using overly complex circuitry.

V. CONCLUSION

The origins and definition of the Least Treatment Principle have been described along with its benefits. Two specific signal processing tasks, compression and expansion, were used to illustrate the benefits of this principle.

The Least Treatment Principle has been shown to be applicable to high-quality signal processing. Processors that follow this principle will yield results which are audibly superior to simpler processors. Because of the lack of side-effects, processing such as signal compression or expansion can be applied in new areas allowing new, creative effects.

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1. R.M. Dolby, "The Spectral Recording Process," J. Audio Eng. Soc., Vol. 35, pp. 99-118 (1987, Mar.).

2. S.G. Cossette, "A New Analog Recording Process for Use With Consumer Recording Formats," 89th A.E.S. convention preprint No. 3004.

GRACEFUL FAILURE: TOLERATING HIGH BIT RATE ERRORS WITH MILD DEGRADATION, NOT SIGNAL MUTING

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ABSTRACT

The current rising demand for high quality digital audio in todays competitive marketplace will require a robust encoding and decoding system for use in noisy transmission mediums. In response to this need, the WavePhore Audio 2000 system employs a unique proprietary digital audio compression algorithm which has proven superior to any other currently available similar system in the industry in transmission links that are not the highest of quality.

1. INTRODUCTION

The WavePhore Audio 2000 employs a unique digital audio compression algorithm which does not mute in the presence of noise but has noise characteristics similar to weak signal FM performance. This algorithm does not require a reacquisition time that is typical of other compression algorithms.

2. HISTORY

In the past, digital audio compression algorithms were all codec with various word lengths. The uniqueness of this algorithm is that it is a bit stream rather than a fixed length codec. The algorithm is based on a unique adaptive delta modulator technique rather than sub-band coding used in other algorithms.

3. NOISE IMMUNITY MECHANISM

In standard algorithms the length of the codec and nature of the codec determines their immunity to noise in the digital data path. A definition of noise is when a data bit is changed from its correct state to the opposite digital state (1 to 0 or 0 to 1). If we imagine a very simple codec which is 8 bits in length and the volume level was determined by the position of the bit in the codec, then if in an algorithm 00000001 represents a small change in sound level and one could see that an error which inserted only one bit in the first position, 10000001, would totally change the volume level of the sound. Since codec are not so simple the resultant effect of the bit error will vary with the codec.

With the complexity of a modern codec, the errors produced by noise could cause the codec to lose its framing information and cause its microprocessor to recognize this loss of framing and cease operation of the decoding process. These codec would require the reestablishment of the framing cycle before they again transmit an audio output. The established codec seem to all fall into the muting mode with error rates of between 1 error per 10,000 to 1 error per 100,000 bits.

The WavePhore Audio 2000 has an algorithm which examines the change in bit stream to develop its audio output. The use of state of the art DSP and DAC technology, in conjunction with some psyco-acoustic principles, enables 16bit accuracy and wide dynamic range at low data rates. The change from one bit to the next determines the output of the algorithm. Because of this nature, the algorithm produces a high tolerance to noise.

4 . NEW ALGORITHM

The WavePhore algorithm is based on an adaptive delta modulator. History has shown that the delta modulator is the most noise immune method of converting analog to digital data. It suffers because of its digitizing error. In the adaptive equalizer, this is overcome by in essence changing the slope or the digitizing steps as the signal varies. Figure 1 shows a mechanism for producing this delta modulation. An algorithm which changes the comparator input produced by the d/a convertor in response to the lack of crossover of the modulator is an adaptive delta modulator. In the encoder shown in figure 1, the DSP contains the algorithm which produces the required effect. The encoder

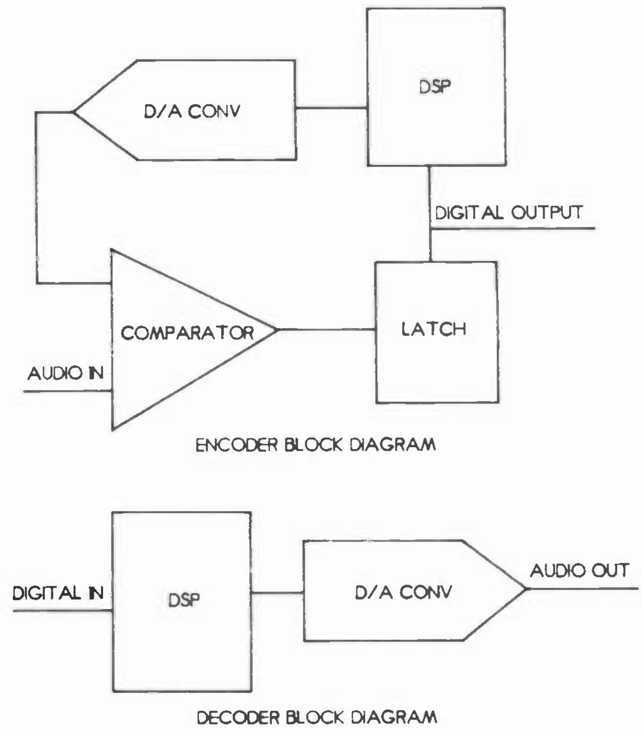

FIGURE 1

signals shown in figure 2 are representative of a operation of the WavePhore adaptive equalizer. They have been exaggerated to illustrate the concepts required here. The decoder signals shown in figure 2 are the analog output signal of the d/a convertor and the digital input signal to the DSP. As shown in the analogue output signal the effect of the noise pulse is a minor glitch and a resulting dc offset of the signal. Provisions are made in the algorithm to prevent an accumulation of d/c offset due to noise from affecting the overall headroom of the decoder. This is accomplished by a secondary algorithm which averages the audio output and adjusts for a long time dc average of zero.

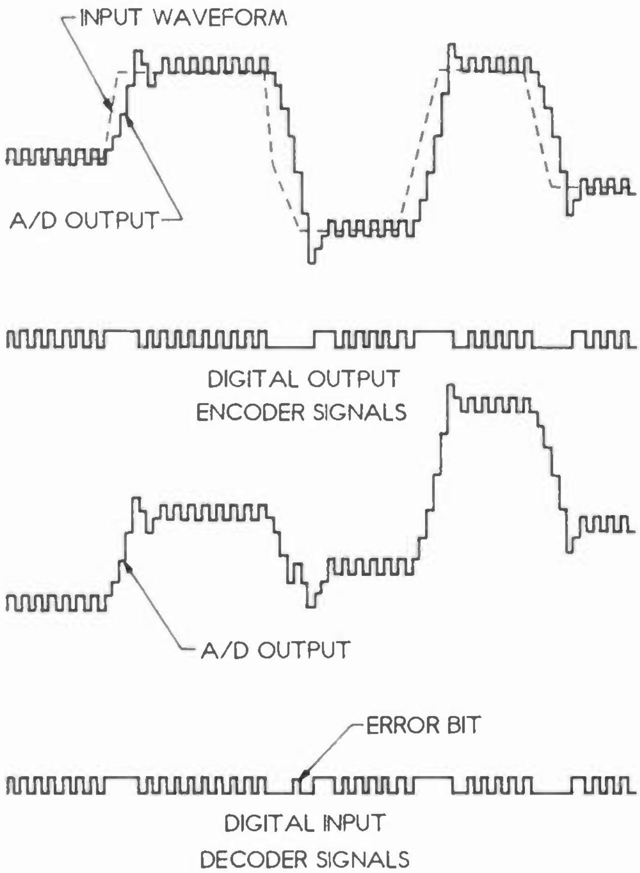

FIGURE 2

This algorithm, because of its very nature of not being word dependent provides the noise degradation improvement. As shown above, the extra noise pulse causes momentary effects on the output. These effects are not audible when the rate is above one error in ten thousand and become audible when the rate is one in a thousand or below. The sound produced is very similar to what is produced by a FM radio signal received in a car when passing under a bridge and at very high bit error rates similar to a very weak fm station. Voice information is intelligent to levels below 1E-2 bit error rate.

5. CONCLUSION

The use of an adaptive delta modulator as a digital audio compression technique is valuable in those areas where the digital channel is not of the highest quality. This technique requires no forward error correction to deal successfully with this noisy environment. Digital links subject to rain fade and other weather effects and other noise inducing problems can successfully be managed by using this technique.

NOTE: Codec is the encoder/decoder algorithms.

Biography

Mr. Mel Engel has been the Manager of Engineering at WavePhore, Inc. since 1991. Mel has been heavily involved with electronic design, consulting and management in an engineering career that has spanned 30 years. He has considerable design and application expertise in areas of digital and analog circuits, power supplies, terminal systems, aircraft navigation and avionics equipment, audio recording equipment, and inertial guidance systems. His degrees include a Bachelors of Electrical Engineering and a Master of Science in Management Engineering. Mr. Engel is a member of IEEE. He currently holds one patent and has a patent pending for Digital to Analog and Analog to Digital Signal Processing.

DESIGNING THE ULTIMATE HIGH QUALITY DIGITAL EQUALIZER

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Abstract.

The advent of the digital audio in the shape of the compact disc took the imagination of the general public, it offered an ease and reliability of use not matched by the compact cassette and an audio quality that unfortunately showed up many of the problems of current studio equipment.

The natural expectation of professional audio engineers was that the performance leap from analogue to digital consoles would show the same stunning qualitative improvement as demonstrated in the change from black vinyl to compact disc. This has clearly not happened; the question is why? This short article is intended to highlight one particular digital console design difficulty and its solution.

Digital Audio Needs.

As products evolve, the technology employed and the increased functionality that was once thought to be spectacular becomes old hat and more is demanded by the customer for the same selling price. Digital Audio is no exception. One such example is the drive to provide total dynamic automation. Indeed the totally automated mixing console is becoming a reality. This requires that all signal processing functions, not just gain, be capable of real time automation. This imposes a number of design and performance criteria on any such system. The most important of which are the need to provide adequate control bandwidth and the ability to calculate coefficients, update processing and gain parameters in real time, without audible artifacts.

It is clear that the primary consideration of any equaliser design must be that of sonic quality. With digital electronics potentially we can copy any existing equaliser design with a level of precision and repeatability not achievable in the analogue domain; the question is, can we prevent unpleasant digital artifacts occurring?

The Digital Eq.

Before we start on digital equalisers let us just remind ourselves of the principle behind an analogue circuit. Figure 1 shows an operational amplifier with a frequency dependent circuit in the feedback loop. If this is a series capacitor with a resistor to ground the amplifier will have a falling response with rising frequency. Changing the RC combination will cause the response to move in frequency but not in slope. Adding additional RC

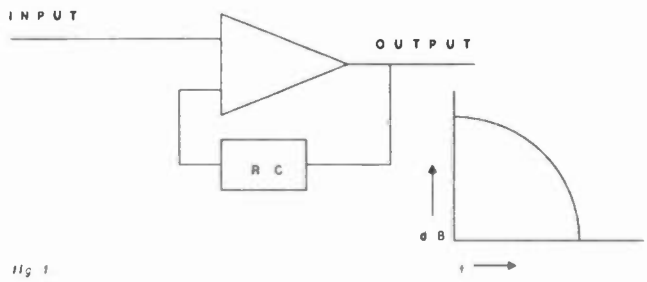

fig 1

networks will cause the slope to change. This is the basis for all frequency dependent circuits.

How can a digital circuit be made frequency dependent; there are no capacitors in the digital domain? All digital systems differ fundamentally from analogue ones in that the sound is broken up into discrete samples. The capacitor has to some degree a memory, it will not change its charge instantly, it is this effect that causes the capacitor to be a frequency dependent component. The simple analogy in the digital domain is a delay. Take the second figure. (over page) We have a simple one sample delay and a two input adder. When we add together the previous and current samples the effect on the audio signal at frequencies many times below the sample rate will be small. This is because the audio level change from one sample to the next is negligible so we add together two similar signals. However as the audio input approaches half the sample rate things begin to happen until at exactly half the sample rate the audio output is exactly cancelled; we therefore have a notch filter with infinite cut off at half the sample rate. If the sample rate is 48kHz (the normal professional rate) then the cut off

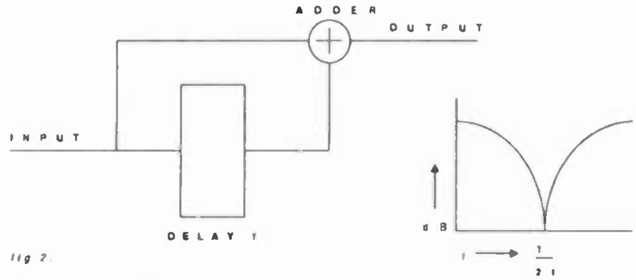

fig 2

will be at 24kHz.

If we now vary the addition ratio of current and previous samples we can vary the depth, but not the frequency, of the notch. In order to change the frequency we can provide a two sample delay we would have a notch at one quarter the sample rate (12kHz), three samples delay one eighth the rate and so on. This is all very well if we want a switched filter at sub-multiples of the sample rate, but is not much use in modern audio console. We can provide a more flexible filter structure by using the output of many sample period delays and adding them together with a variable gain. This is known as a Finite Impulse Response filter. However it suffers from one fatal flaw that makes it unusable in a console; the time taken for the audio to travel through the delays which could be several 10's of milliseconds.

A more practical form shown in the next drawing (fig 3) is known as the direct form of an Infinite Impulse Response filter and the most obvious feature is the relative simplicity of the hardware. The signal from the delay taps is fed back into the input; careless multiplier

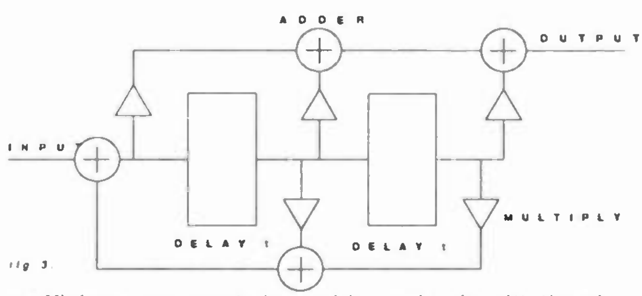

fig 3

coefficients can result in positive gain- it will 'hoot' or latch up, furthermore recycling the audio data may cause truncation errors to accumulate causing loss of resolution. However this structure can be made to provide the types of responses we are used to in the analogue console world.

This is therefore the basis on which digital equalisers are constructed. The resistors and capacitors of the analogue circuits are replaced by multipliers and delays.

The DSP.

As long ago as 1978 Neve started work on digital signal processing. In those days much of the work was of a fundamental research nature, 16 bit multipliers were only just available, 16 bit quality A-D convertors a pipe dream. However work that was undertaken, in collaboration with the BBC, which indicated that these 16 bit multipliers would not provide adequate audio performance. We therefore developed a system using floating point arithmetic based on a 16 bit mantissa and 4 bit exponent. We all assumed that the theoretical dynamic range of 144db would be such that the sonic quality would transform the audio industry. Sadly this was not the case, mainly due to the ear's ability to hear non linear effects 60 or 70 dB below the audio, in short, operators were not happy to lose sonic quality at the expense of operational convenience. Evolution to 24 bit hardware offered improved resolution and indeed the need for 24 bits was proposed prior to the availability of suitable DSP devices, which are now found in many applications.

It soon became apparent that even the fixed point format of 24 or 32 bits was not sufficient, especially in large scale music recording and mixing.

Thus many designers turned to IEEE floating point processors, which offer 24 bit resolution but with hundreds of dBs of dynamic range, through an 8 bit exponent. These processors are aimed at the world market for scientific computing applications, but are usable for some digital audio applications. The dream was that limit cycles, d.c. offsets, frequency domain errors etc, were now a phenomena of the past. It was therefore natural to port previously known algorithms into the floating point domain with the assumption that these previous unpleasant artifacts were now below the source input noise levels due to the increased dynamic range of the floating point hardware.

This paper looks at the performance issues of 4 types of filter structure from a console system point of view, when implemented using the IEEE floating point format, with some interesting results.

Filter Structures

The analogue world has evolved over the years many different types of filter structure, well known examples are Tow Thomas, Baxandell and Sallen and Key. The digital world is just the same, there are many filter structures and new ones appear regularly in the scientific journals.

In order to establish the suitability of a filter structure all theoretical aspects must be explored and the performance simulated or measured against a reference. These filter designs have been evaluated using theoretical research, followed by simulation and physical implementation. The physical implementation is necessary to finally establish the dynamic issues such as ramp rates and the filter time domain dynamics, along with audio quality.

Four filter structures were investigated;

Transposed Direct Form (TPDF)

This structure is similar to the direct form that was shown before and has been adopted by many manufactures because it has been well documented. The noise performance and coefficient sensitivity still degrade with falling frequency, however the interesting factor was whether this would now be improved as a result of floating point processing to allow the filter structure to be used across the audio band in systems which will demand 20 bit system performance.

Z hat Form

A variant of the TPDF structure, which has been described in a number of papers. The structure arises from modifying the TPDF. However the noise performance at high frequencies degrades and the question was whether by using floating point processing this structure would offer the required performance.

Neve N1 Form

Research was performed to obtain a filter which had the improved coefficient control but which would avoid the problems of both the Z hat and TPDF forms at extremes of frequency in terms of noise and frequency domain errors resulting from coefficient quantisation.

Neve N2 Form

A structure which is a further developed version of the N1 Form and provides improved low frequency noise and D.C. performance whilst retaining all other attributes.

Test Environment

All filter structures were analysed theoretically and then developed further using a computer simulation and measurement platform. This allowed various test signals, with and without dither, to be generated for the structures under test. Finally the structures were implemented on hardware to test the audio sonic performance.

The simulation environment used 64 bit arithmetic (configured as 53e11) as the format for the reference filter and test signal generation and analysis. The filter structure under test was implemented in 32 bit (24e8) arithmetic.

Two basic tests were carried out.

Test One

A 2nd order filter was set to +12dB with a Q of 2 and the frequency setting swept from 0 to 20kHz. The test tone was a 0dB sine wave at a frequency in the filter's transition band. The RMS errors, against frequency for each filter structure are shown in Fig 4.

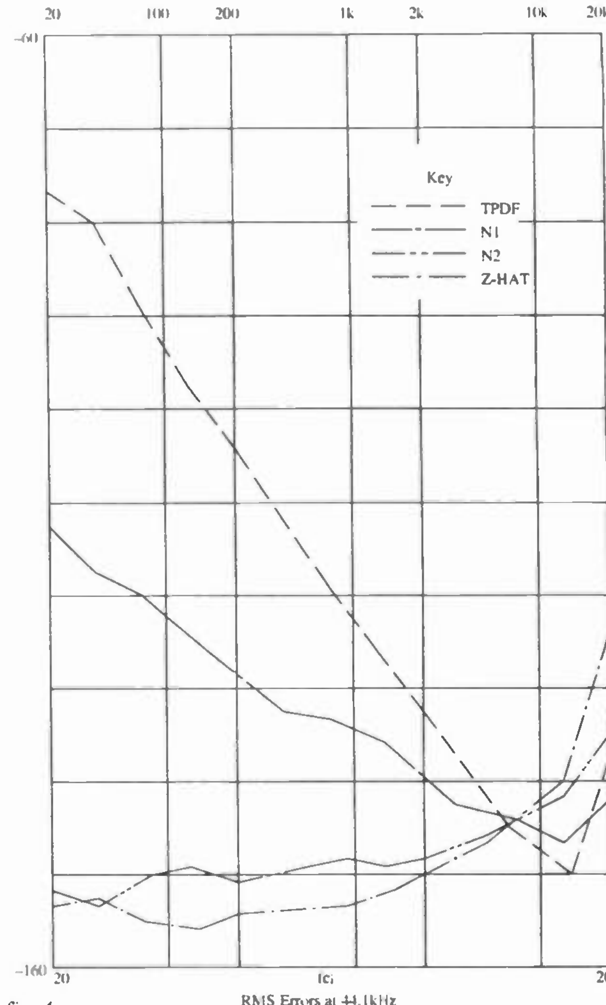

fig 4.

TPDF - The results clearly show poor low frequency characteristics. Also a major D.C. offset was found to exist. The performance is adequate at high frequencies but the structure does not clearly meet the performance criteria,

Z-HAT - Here the results are very good at low frequencies and indeed at higher frequencies the filter is usable. However the noise increases sharply at the edge of band and may give rise to unwanted imaging affects.

Neve N1 Form - Results are better than the TPDF but still not as good as the Z- HAT at low frequencies. However the D.C offset was much reduced.

Neve N2 Form - Performance much more consistent across the frequency spectrum. The D.C. error was minimal.

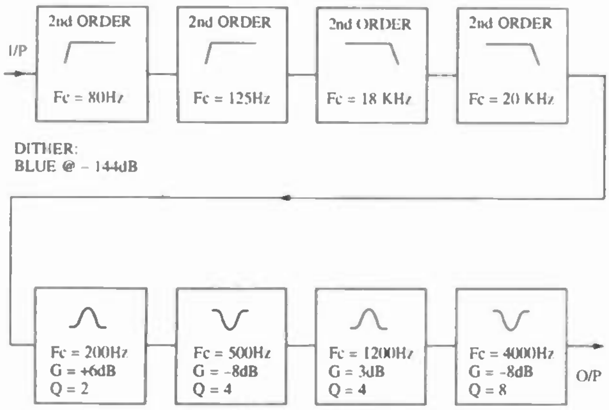

fig 5. Filters For Test 2

Test Two

This time a complex set of filters was cascaded in order to represent a practical console path. (fig 5 above). The test input source was a wideband signal made up from sinusoidal odd harmonics at 1kHz, 3kHz, 5kHz, 7kHz, 9kHz and 19kHz, which was then dithered at the 24 bit level. It is important that measurements are made with

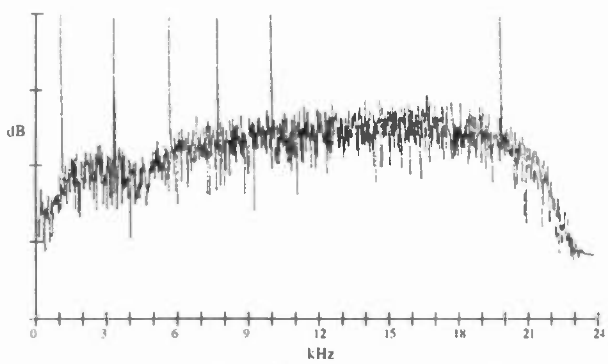

fig 6. Reference filter (53e11) filtering a dithered signal at 44.1kHz

low level modulation. Truncation and co-efficient control errors are not evident without signal. It is the low level modulation errors we are interested in. The system was simulated using the 64 bit (53e11) as a reference, at a sample rate of 44.1KHz. The output being measured using an FFT analysis. (fig 6.)

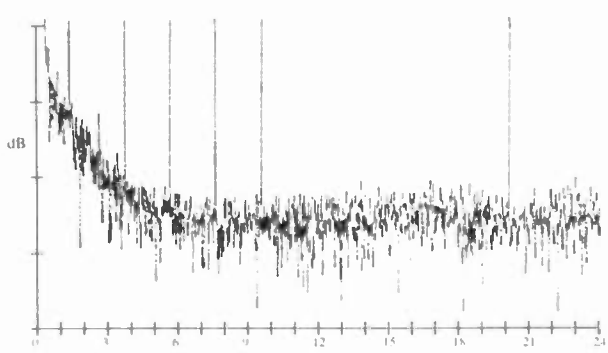

fig 7. Text book direct transpose (24e8) filtering a dithered signal at 44.1khz

If we now compare the 64 bit reference system (fig 6.) with the 32 bit structures we gain further insight into what is happening.

Firstly the 32 bit TPDF structure (fig 7.) shows a large rise in noise at LF and a failure to fall at HF. We should have shaped our input noise with the low pass filters at 18K and 20K. RMS noise is some 35 dB worse than the reference, notice a peak at 2kHz.

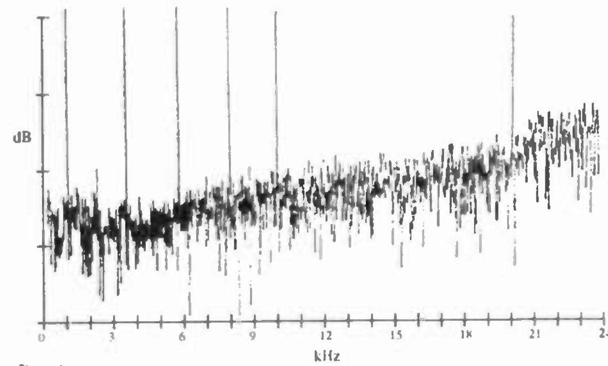

fig 8. Z-hat filter (24e8) filtering a dithered signal at 44.1khz

Secondly the Z-Hat form (fig 8.), here the problems are severe at HF, notice again the peak at 2kHz. Overall noise is now only 9.1 dB from the 64 bit system.

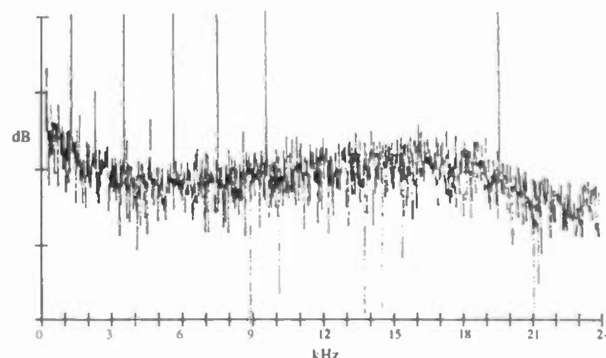

fig 9. Neve type 1 filter (24e8) filtering a dithered signal at 44.1khz

Next the Neve 1 form. (fig 9) We have a better fall at HF and a less pronounced rise at LF than the TPDF.

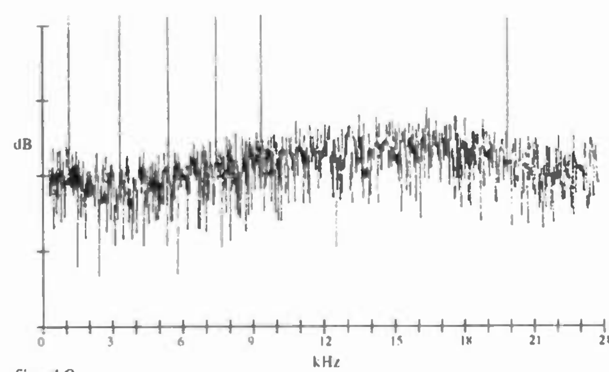

fig 10. Neve type 2 filter (24e8) filtering a dithered signal at 44.1khz

The Neve 2 IEEE form (fig 10.) nearly matches the 64 bit reference to within 1.1 dB and the 2kHz peak has gone. All these structures were implemented using 32 bit IEEE floating point arithmetic with a theoretical dynamic range of 1500dB. However by the use of our own ASICs we are able to use our own word length formats optimised specifically for digital audio. We have developed a chip set allowing us to go even better than the IEEE N2 form and are able to get within 0.1dB of the 64 bit reference system.

Conclusion

Clearly the expected solution to the performance problems with the original 16+4 designs is not solved just by throwing more bits at it. The TPDF structure not even managing a 16bit theoretical level even though 32 bit IEEE floating point arithmetic is used.

This short paper has shown the need to analyse carefully the choice of filter structure required for high performance digital audio systems. The truth is that there is no easy way to create professional digital audio equipment: both hardware and software solutions require great ingenuity and painstaking analysis (both analytical and human) if they are to truly succeed.

DIGITAL AUDIO STORAGE

Sunday, April 18, 1993

Moderator:
Fred Morton, KMGZ-FM, Lawton, Oklahoma

PRACTICAL STATION EXPERIENCES WITH DIGITAL AUDIO STORAGE
Rick Fritsch
KBZQ-FM
Lawton, Oklahoma

KEY CONSIDERATIONS WHEN CHOOSING A HARD DISK BASED DIGITAL AUDIO STORAGE SYSTEM
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Broadcast Electronics
Quincy, Illinois

SPECIFYING DIGITAL AUDIO WORKSTATIONS FOR BROADCAST
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Ty Ford
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PRACTICAL STATION EXPERIENCE WITH DIGITAL AUDIO STORAGE

Rick Fritsch
KBZQ-FM Radio
Lawton, Oklahoma

Audio storage and playback are undergoing dramatic and rapid changes at today's broadcast station. This is one opinion that says the changes are for the better and improvements are developing almost daily.

BACKGROUND

When planning a new broadcast facility, I made it a point to plan to NOT having any cart machines of any type in the facility. I had decided to affiliate with a satellite service to provide programming for the new station and in that decision I knew an "automation" system was needed. I had been familiar with the typical automation system with lots of carts, carousels, go-carts and reel to reels to store commercials and/or music. This technology based in the sixties and seventies was not the answer for a new station going into the nineties. What was needed and discovered was a system that provided instant access to hundred's of commercial cuts, and liners that need to play on a satellite network.

Shopping

I searched the fall NAB convention in San Francisco in 1991 and the spring 1992 Oklahoma Broadcasters Convention, read every advertisement available to narrow the growing field to a handful of competitive priced systems. Shopping proved to be quite an undertaking as specifications and design philosophies varied greatly from one vendor to another.

Performance Specs

The system had to be able to perform reliably with up to three days walk away time, switch multiple sources, be easy to learn; since a number of employees would need to be trained quickly over the years; and have a low maintenance schedule. The audio had to be excellent and some features to be included in the system were items such as silence sense, automatic replacement of out-of-date spots with programmed fill material, provide audio overlap, be flexible enough to accept the format and requirements of today and tomorrow's needs without complete replacement. The thought here was to provide a system, that could switch from satellite provided music to music provided from CD jukeboxes. The system had to have enough storage space to store commercials for even the busiest time of the year and store over two thousand liners for the satellite, without the need to delete information because we were running out of room to store commercials.

Configuration

The system I decided to purchase was from Systemation, Decatur, Illinois. This system is housed in a personal computer cabinet with a small footprint, a CRT and keyboard; the only difference from a regular PC being the ports on the rear of the cabinet to connect to the terminal board for the audio input/outputs and the satellite commands. The system came equipped with inputs for the satellite network, an external news source, and the hard drive audio source. The system was quickly updated with another audio card to accept another outside audio source to do unattended remotes with our RPU system. This is the system in use today. It stores over 400 minutes of audio material, in mono, and has instant access to any of the cuts in milliseconds. The system passes stereo audio from the satellite and other external sources, the only mono source is the audio from the hard drive.

INSTALLATION

Connections

Connections to the system are made extremely easy by use of pressure screw contacts on a board that connects to the CPU with ribbon cables. This made the connections easy and very fast using tinned wires only. The Systemation approach is to use high level unbalanced connections to the system, but the instruction book accompanying the system leads you through the connections and lets you use balanced lines in conjunction to the unbalanced lines of the Systemation system. The commands from the satellite receiver are made with a multi-conductor cable from the satellite receiver to the circuit board near the connection for the audio.

Testing/Set-up

I found the system easy to connect and the testing that followed quickly led into actual use. The basic testing consisted of recording and playing back audio from the hard drive of the computer and then confirming that the commands from the satellite triggered the proper events to play in the system. The system came pre-loaded with most of the configuration of the Jones Satellite network we carry. Further programming made some additional changes to the system to customize the system to our station application.

ACTUAL ON-AIR USE

Recording

Recording is an easy process with the Qwik-Disk system. The programming makes use of pull down menus to choose a function and execute, much like a Macintosh operation system. We first recorded the liners sent from the network, then generic promos and what few commercials we had in the beginning. The system easily handles the recording of the material; the operator is familiar with the recording system in only minutes. In repetitive recording functions, the computer remembers the number of the last track you recorded and lets you choose that track or another, then lets you fill in the information for the track and finally remembers the length of the last cut so that in repetitive recording situations you do not have to reset the time again and again. Recordings made can be checked by playing back information in an off air situation from the "production" playback channel. You may even edit your production as to end date, name or title of the track and the length of the track. This is extremely helpful when you don't know the exact length of the track to record; you can set the length for longer than you need and then later edit the time for the exact length of the track.

Other features are too numerous to mention, but let you customize a rather generic system to you particular application. The system must be DESIGNED as to the number of inputs you wish to have for your system from the factory, or you may specify a full complement of the input sources for present and future use. The rest of the customization comes from software changes, not from additional hardware.

Loading of Program Log

Our program logs are generated on the CBSI Traffic and Business system and manually loaded into the Program Automation. Most hours of the day are the same as far as the program log of commercials and do not change with regularity. Commercials and other short elements are loaded into the program automation in less than twenty minutes.

Sports represented a unique problem to play music bumpers after each commercial break going back to the play-by-play action. This too was addressed by the system and handed fairly well. Due to the limitations of the design, the source dedicated to the music bumper could not be accessed if the system were to be put into a paused situation; as it often occurred when airing play-by-play sports. We were able to work around the problem most of the time.

The program log is manually loaded into the program automation, but the software could be purchased from CBSI to have the system download the commercial log to the Systemation via floppy disc. We have chosen to retain manual entry, as we have to write on a few commercials from the Unistar Ultimate network to fulfill our obligation to the news network, when not airing the news on a real time basis. We "customize" the log in a few instances to move commercials to better fill the breaks provided by the music network. We also have to fill breaks that are short from the pre-printed program log that do not completely fill the satellite audio's window for a spot break.

Maintenance of Hard Drive

While the complete automation system is a far cry from the large wall of racks that I worked with even just a short five years ago, the system DOES need some maintenance. The maintenance is more software related than hardware related. The provision must be made for when the hard disk crashes and destroys itself. Note I said when, not if, because it is only a matter of time when the disk will wear out and need replacement. When replacement becomes necessary, the fastest way to replace the data on the disk is from a reliable floppy or tape backup. To backup the 435 megabyte disk to floppies would require 232 floppies and over four hours to backup the hard drive. This is more quickly accomplished by doing a backup with tape drive, each tape holding 60 megabytes of information. The process still takes over two hours, but protects you for the time when you will need that backup to replace the hard drive when it crashes. This full backup is accomplished once per week on Sunday night after 10 pm, when there are no commercials scheduled. The system will not at present do a full tape backup and perform the routine duties of playing audio from the hard drive. The simultaneous job of backing up and functioning to playback and record is still in the future, although the near future. The other job that needs to be accomplished is to unfragment files written to the disk over a period of one or two weeks to speed the operationg efficiency of the hard drive. This process utilizes one of the speciality programs such as PC Tools or Norton Utilities. Both perform the task of grouping the files together for more rapid access of the hard disk head to read information for audio playback. This unfragmenting of the hard drive takes only about twenty (20) minutes and is also performed on Sunday nights.

SERVICE

The service provided from the vendor has been great, with a phone call, they would call back on their outgoing WATTS line and stay on the line until any problem was corrected. Any seller of equipment without a full time service department available 24 hours a day can't hope to compete in the marketplace. When a problem occurs, it is often necessary to call at odd hours or on weekends to discuss problems and get solutions.

COMMENT

The system has performed well over the past year with a few notable exceptions. The system had a habit of hanging up and not moving to the next event in the middle of playing back hard drive audio. This problem was quickly fixed with a software update provided by the company. The other problem is that very occasionally the system the system will "hang" on the hard drive like a compact disc that hangs on one particular spot on the disk. This problem has only occurred two or three times, and at present is not regular in its occurrence. This system does not rotate commercials for clients that have multiple cuts to air within one schedule. That process must be a function of traffic. While the concept is somewhat unfamiliar to previous operations I've been associated with, it does allow a better control of which particular cut aired at what time, as the traffic computer keeps a better record than can be had by simply saying all the cuts rotated equally on the schedule. Other manufacturers offer this feature, I believe Systemation's approach is one that is well thought out and will in the end provide the best information if questions are raised about a multiple cut scheduled are raised. This system also has a limitation in that only two "channels" are available to play or play/record information to the hard drive. If you are doing production, recording a commercial or listening back to a commercial or listening back to a commercial just produced, that leaves only one channel with which audio can be sent over the air from the hard drive. At that point, you lose the ability to do overlaps of commercials during playback. We get around this by dubbing commercials into the system before and after the spot breaks play. After getting used to this limitation, we find some restrictions in doing the production, but one that can be managed. Other systems I shopped did not have this limitation, but the difference was in price. Others offered file servers or mirror systems in production and on-air, but prices were half again as much to double the price paid for this system. The overall operation of the system would rate 97 percent in operation since we signed on with the system. Audio is always very good and when the operator dubs the material into the system properly, the on-air sound is great. There is another Systemation system in the market operating, and the air sound from the system also is very good and is very reliable. I would recommend that when choosing a system, have firmly in mind the tasks you wish to accomplish and design a system specifically to do those tasks and have the ability to change with your changing situation in the future.

This will make the investment last many years beyond the norm to provide excellent return on investment.

I believe that in the near future most all stations will be replacing the familiar cart machines with a system similar to the system now in use at KBZQ. This will allow the better control of commercials and music aired, and even allow stations in major markets to "automate" certain parts of the day or week to increase efficiency

KEY CONSIDERATIONS WHEN CHOOSING A HARD DISK BASED DIGITAL AUDIO STORAGE SYSTEM

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ABSTRACT

Today's Digital Audio Storage Systems are, at a minimum, replacements of yesterday's analog cartridge machines. But they can be a lot more than that as traditional cart machines have never before effected overall station operations as these systems can. The Digital Audio Storage System provides stations great latitude in making the best choice for their operations, ranging from a simple cart machine function to a multiple station, totally integrated system. The process of selecting the best system is complicated because it may be prudent to change existing station operations. Before deciding which type of product to purchase, it is necessary to plan with an eye to the future for overall operational changes and to plan for the Digital Audio Storage System's arrival. This paper is intended to assist station personnel in identifying the key considerations for the successful integration of a digital audio storage system in their station.

WHY CONSIDER A DIGITAL AUDIO STORAGE SYSTEM?

Productivity / Operational Enhancement:
The Digital Audio Storage System should provide the means to improve the flow of work processes within the station without adversely affecting the station's on-air product. Operation of multiple station facilities could be done with fewer staff, yet provide an "On Air" sound that is tight and reduces human error. The recording of audio material can be done in a central studio with distribution of the digital audio to the appropriate "On Air" studios. The Digital Audio Storage System should also provide for the simultaneous operation of multiple, independent "On Air" and production studios without creating conflicts during the audio retrieval process.

The Digital Audio Storage System will improve the sound and management of a station's audio inventory by reducing the requirement for lossy analog audio reproduction. The audio quality of digitally reproduced material sounds as good from the 100th generation of playback as it did during the first since there is no analog tape to deteriorate the sound. The recording of audio material is simple and will no longer require a cart inventory nor traditional "cart" operations such as bulk erasing and splice finding.

Operational Expense Reduction:
The Digital Audio Storage System will improve the efficiency and increase the output of the operation's staff mainly by reducing the number redundant tasks they have to perform. This improvement in staff output will present new opportunities to assign the personnel to take on "revenue generating" tasks and increase the operation's financial success. The Digital Audio Storage System will also reduce material and maintenance operational expenses such as carts and associated cart machine parts. Depending upon the operation, the Return on Investment for a Digital Audio Storage System could be less than two years.

PHASED PLANNING

System Requirements:
Careful consideration of the intangible aspects of an operation must be taken into account. These aspects include defining the operation's current needs as well as future growth. Other aspects include the purchase,

installation, training, and system support requirements. This multiple step system planning will encompass the entire process and lay the groundwork for the successful implementation of the Digital Audio Storage System.

Design Planning:
Analyze the operation's current and future growth plans. How many positions require simultaneous operation? Is the station looking for simple commercial storage, satellite automation, or live assist automation? What kind of upgrade path will the station's operational environment require? What is the requirement for each individual station when looking at multiple station operations? What other special requirements will your operation need to have fulfilled? Write this information down on paper and use it as a map to form the cornerstone of your system design.

Product Information Collection:
Begin collecting information about applicable products. This can be accomplished through attendance at trade shows (such as NAB), gathering product literature, visiting current user stations, and visiting the Digital Audio Storage System manufacturers. This process will enable the planner to virtually see and touch every applicable Digital Audio Storage System possible in advance of making a purchasing decision.

Staff Implementation Planning:
A Digital Audio Storage System will drastically change the way an operation creates their On Air product. Careful planning must be done on how the Digital Audio Storage System will impact the operations as a whole and how to prepare the station personnel for these changes. The staff will have to learn a new system and computer operations. In addition, the staff will no longer have the traditional security of a cart in hand, arrange carts sequentially for On Air playback, or have to put them away. All of these processes, which were once part of the "creative routine", now become a "matter of trust". The staff must be confident that the Digital Audio Storage System will have the audio correctly arranged for playback and play it when requested. In a high pressure environment, where emotional creativity is the norm, uncorrected playback errors, whether human or product induced, could escalate into much larger problems like staff frustration or rejection of this new technology.

The station staff can also resent a product which could possibly reduce or eliminate their importance at the station. Be sure to be open with the staff on the features and benefits of the Digital Audio Storage System, and why you are adding this capability to your operation. Explain their role with your operation and how they can positively impact the successful implementation of the product. If the fear or suspicion of the system is too strong, excessive complaining or even sabotage of the system's operation could occur. Motivate your staff positively - it is human nature to fear what we do not understand.

Plan to Manage Change:
Don't change too many aspects of the station's operations at once. Trying to switch to a Digital Audio Storage System and/or program automation at the same time you're changing a format or traffic & billing system could be "information overload" for the staff. This type of overload will cause stress and further reduce the ability to carefully think out problems and implement corrective action. Use a step by step process with personnel assigned to do particular tasks. Plans which establish responsibility and anticipate mistakes will smoothly integrate change into your station's operations.

Plan for a Station "GURU":
It is absolutely necessary to identify and train a station GURU who will be the champion of the installation, training, and phase over to the new Digital Audio Storage System. You can usually find one person in the station who really wants the product, has the desire to learn computers, and will be a responsible "lead person" for making the purchase a financial success. Be sure to allow training time for this GURU, whether it is making time for the employee to attend product training or have the time to learn off-line how

the system operates. In an emergency, one cool thinker who has a comfortable knowledge of the system will be able to do whatever is required promptly and efficiently.

Plan for "Parallel Operation":
Completely dismantling the existing operational system before the initial problems and modifications of the Digital Audio Storage System have been worked through carries high risk. Plus, the possible errors resulting from this "forced through" approach can cause high amounts of staff frustration from the pressure of learning a new system while trying to correct its operational errors. The best approach would be to have the system operate off line in parallel with the On Air product. Run the system for a week or two, train the staff, uncover the problem areas, and then issue corrective action. You will find that Parallel Operation will minimize the risk the station undertakes when installing a Digital Audio Storage System.

Once the system is on line, maintain the capability to switch back to the old system (or portions thereof) for a couple of weeks. This switch back capability will provide some level of comfort to personnel, and is an effective back-up if the Digital Audio Storage System experiences "infant mortality". After the initial startup period, it is wise to have a back-up plan prepared on what to do during the various types of "emergencies" which can occur. These emergencies could occur as a result of Digital Audio Storage System hardware, software, or hard drive failures... plan with the knowledge that the worst disasters are those which are unprepared for.

DETERMINING STATION REQUIREMENTS

There are several key parameters to be aware of when selecting the proper system. These parameters are separate from the system requirements and actually address how the audio is stored and distributed. You should determine what effect, if any, these have on your "concept" system.

- Is it multi-user? How many users can I have on the system at the same time?
- Is it multi-tasking? Will I be able to distribute other pertinent information on the system while using it for playback or recording (i.e. Electronic Mail, ASCII text information, Call Logging, etc.). If I can use multi-tasking, can I import standard text files or must I re-type all this information into the system by hand?
- How much storage time will be added to the system? What is the maximum?
- What types of sampling rates are available?
- Is the audio uncompressed? If not, what types of compression are used? Will I be using other types of compression in the audio path?
- What types of outputs are available? What is the data transfer bandwidth over a network (ie. Novell or Lantastic)? Identify examples of transfer times for :15, :30, 1:00, 3:00 and 10:00 minute programs on a busy network. If a network is used, does it use a data compression scheme?
- What types of remote control functions are available for frequently used commands such as PLAY, STOP, PAUSE, or RECORD? Can other commands be accessed by external switches?
- Can I use the system without a keyboard? How is this accomplished?
- Will I need satellite, CD, R-DAT (or any combination thereof) types of automation? Is it available and how is it done? What models of playback devices are compatible with the system?
- What types of external machine control are available? (ie. random access, simple start / stop)
- Is hard drive mirroring available? How is it done? What happens if a drive fails?
- What types of personal computers are used? Can I supply my own? What happens if a computer fails?
- What else can go wrong? What mechanism is built in to identify and correct mistakes?
- How will I be able to integrate traffic and billing files into the system? What about music logs? News Wires? Local Area Networks?

These parameters are a few of the basic questions which should be answered with the consideration of a system. This type of preliminary analysis could mean the difference between getting out all that is required of a system or being "stuck" with a system which only meets selected current requirements.

It is beneficial to describe, on paper, your system requirements in a point by point fashion. This document will serve as the description for your entire system and can be sent to your selected Digital Audio Storage System manufacturer for a response. In this manner, your requirements for the system are clear among your staff and the manufacturer. The manufacturer should also be able to provide some type of point by point response clearly identifying their understanding of your requirements and how their system will comply / not comply with your requirements.

SELECTING A SYSTEM

Once you have identified your station requirements, evaluated the various products in the market, and weighed all the parameters together, you are now ready to finalize your decision. A few remaining questions need to be answered as a result of the following statement:

SOFTWARE IS NEVER FINISHED

With traditional mechanical audio gear, usually what you purchased is what you were using for that product's lifetime. But now that most of the Digital Audio Storage Systems are software based, what you use and how you perform certain functions can change considerably -- even over one year's time. Software is a constantly evolving product, and your investment in a Digital Audio Storage System will be enhanced by a manufacturer's superior software support. Also keep in mind that the specific hardware used by the software of a Digital Audio Storage System is not compatible with other systems. For example, you can not use Brand A's software with Brand B's hardware. Changing over to a different system at a later date can cost the station both time and money. This initial choice of which system to use is an important decision regarding your future operations. Keep in mind that when you are choosing a system from a certain manufacturer, you are in effect signing a business partnership which places responsibility for your system's growth in the manufacturer's hands. You can evaluate the following criteria to help select which system you purchase:

- How long has this manufacturer been in business? What is its main line of business? Has it been successful?
- Does the company have a strong base for product and software support?
- What type of longevity is estimated for the product, its technology, and the company?
- What is the company's track record with previous products?
- How do current users of the system like the product? How do they rate the service and support?

Although the above criteria will not guarantee the success or failure of the system purchase, it will provide useful background information which could indicate future problem areas.

SUMMARY

Your objective answers and follow through to this paper's questions and informational guidelines will put you a couple of steps ahead towards successfully integrating a Digital Audio Storage System in your station's operations.

SPECIFYING DIGITAL AUDIO WORKSTATIONS FOR BROADCAST

Bruce Bartlett
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When you specify a DAW for broadcast applications, you'll have a lot of systems to choose from. This paper will help you choose the right system for your application.

Station-automation system or DAW?

A station-automation system automates the playback of program material, while a DAW is used mainly for editing and playback of a series of spots. A DAW can replace cart machines.

A station-automation system controls playback from several devices (including CDs). A DAW controls playback only from a hard disk.

A station-automation system has a radio-station user interface, while a DAW has an audio user interface.

There is some overlap in functions between the two systems. For example, some automation software offers editing. Most digital editors include a playlist for automatic or cued playback.

Computer-based workstation or dedicated workstation?

A computer-based workstation is a DSP card and software that install in your computer. A dedicated workstation is a control panel with a built-in computer, such as AKG DSE7000, Lexicon Opus, and Korg Soundlink.

The computer-based workstation costs much less than the dedicated workstation. However, a dedicated workstation is easier to operate. That's because it has real controls, and because you don't need to understand computers to use it.

Computer platform: IBM, Macintosh, or Amiga?

DAW systems are available for all three types, but IBM clones cost less and are the most popular. IBM software is available in Windows and DOS versions. Windows software is easier to use for some people, but requires extra memory (on hard disk and in RAM) and a faster computer than DOS software does.

A/D, D/A converters included or not included?

A DAW with its own internal A/D converter is ready to record analog sources. In most current models, the converter quality is very good. If a DAW has no converter, you use the one in your DAT recorder, or use a high-quality external converter.

Most DAWs without converters cost less. Some DAWs with a built-in A/D converter require you spend extra for a card that adds digital inputs.

Definitions

Before recommending systems for specific applications, here are a few definitions.

Tracks vs. channels -- A *track* is recording of a single type of program, such as a narration track, a sound-effects track, or a music bed. A *channel* is an output.

A *virtual track* is a storage area on disk that simulates a tape track. Some DAWs can record several virtual tracks, which can be assigned to two or more output channels. You might assign one track per channel, or several tracks per channel.

A *real track* is a track that has its own dedicated output channel -- one track per channel. A real-track system and an external mixer let you do mixes instantly and change the mix in real time. So does a DAW with multiple output channels.

Many DAW companies call channels "tracks," which adds to the confusion.

2-track DAW vs. multitrack DAW -- A *2-track DAW* is used mainly to edit an already-mixed master tape. A *multitrack DAW* is used to create mixes for spots, etc. The 2-track DAW costs less.

In order to do a mix with a 2-track DAW, you must build up the mix one sound at a time, recording new sounds on top of previous mixes. This is slow and cumbersome compared to mixing with a multitrack DAW.

2-channel DAW vs. multi-channel DAW --
A *2-channel DAW* has two outputs (left and right). A *multi-channel DAW* has four or more outputs. With a 2-channel DAW, mixes are done inside the computer. The computer takes time to calculate the mix. With a multi-channel DAW, you can mix the output channels instantly through an external mixer. Two-channel costs less.

Suppose you specify a 2-channel workstation that has 38 virtual tracks. That means you can assign any combination of the 38 virtual tracks to the 2 output channels. The computer will take some time to calculate the 38-to-2 mix. You won't hear the mix until this is done. If the mix needs changes, you can change the level of any element of the mix, and do another mix calculation.

Suppose you specify an 8-channel workstation. In this case, you might assign a different virtual track to each channel. Once this is done, you can mix the 8 tracks instantly with an external mixer. Or you can mix them internally in the computer, after waiting for the mix calculation.

Tape recorder vs. disk recorder -- A recorder is not really a workstation, but a recorder may be a viable alternative. A *multitrack digital tape recorder* records on tape; a *multitrack hard-disk recorder* records on disk. Tape costs less. With tape, you cannot easily edit or time-shift indvidual tracks. With disk, you can. Disk has random access.

The Alesis ADAT records on S-VHS cassettes; the Tascam DA-88 records on 8mm cassettes. The Anatek RADAR records on internal hard disks. The RADAR is a standalone, multitrack hard-disk recorder. Computer-based DAWs also record on hard disk.

Choosing the right system for your application

*If you just want to edit an already-mixed master tape, get a 2-track DAW. Examples: Turtle Beach 56K, Digital Audio Labs Card-D, Software Audio Workshop by Innovative Quality Software for the Card-D DSP card.

*If you want to create mixes, but your budget is under $2000, get a 2-track DAW. Mixes will be slow and cumbersome to do (like sound-on-sound), but you'll save hardware costs. Examples are listed above.

*If you want to create mixes more easily and can spend more than $2000, get a multitrack DAW. This can have 2 channels or more. Any DAW that is not a 2-track DAW is a multitrack DAW.

*If you want to do instant mixes with an external mixer, get a system with 4 or more output channels. Four channels might be adequate for mono mixes. Eight channels is adequate for stereo mixes, which might include a stereo music bed, stereo effects, and two announcers.

*If you want to do instant mixes, and your budget is under $4500, and you don't need to edit or time-shift individual tracks, get a multitrack digital tape recorder -- Alesis ADAT, Tascam DA-88.

*If you want to do instant mixes, and you need to edit or time-shift tracks, get a multi-channel disk-based system. This system can be one you install in your computer, or it can be standalone, such as the Anatek RADAR.

*If you want to do mixes, and you need to edit or time-shift tracks, and your budget is under $4500, get a multitrack, 2-channel DAW. In this system, the computer calculates the mix internally, rather than you doing a mix with an external mixer.

With most multitrack DAWs, you can assign only one virtual track to each output. To add more tracks, you must bounce tracks first. In contrast, some multitrack DAWs let you mix several virtual tracks to each output, which is faster. Examples are MicroSound and Studer Dyaxis.

Some desirable features to look for in a DAW

*Fast soundtrack redraw. Look for software that writes a soundtrack graphics file to disk when you make a recording. In this case, when you change the view of the soundtrack, it redraws quickly. In slower systems, whenever you change the view of the soundtrack, the computer must recalculate the soundtrack from audio data on disk.

*Time compression/expansion. This adjusts the length of a program to fill a certain time slot.

*Time shifting of tracks. This allows precise timing of audio events in a mix.

*Level and EQ adjustments. These allow you to match the volume and tone quality of various audio segments, duck music under voice, and so on.

*Fast, non-destructive deletes. Destructive deletes re-write data on the disk, so they take a long time to calculate. Non-destructive deletes change pointers to data on the disk, so they are almost instant.

*Automated mixing. The computer remembers your mix moves, such as level adjustments for each track. Probably this is necessary only for 8 tracks and up.

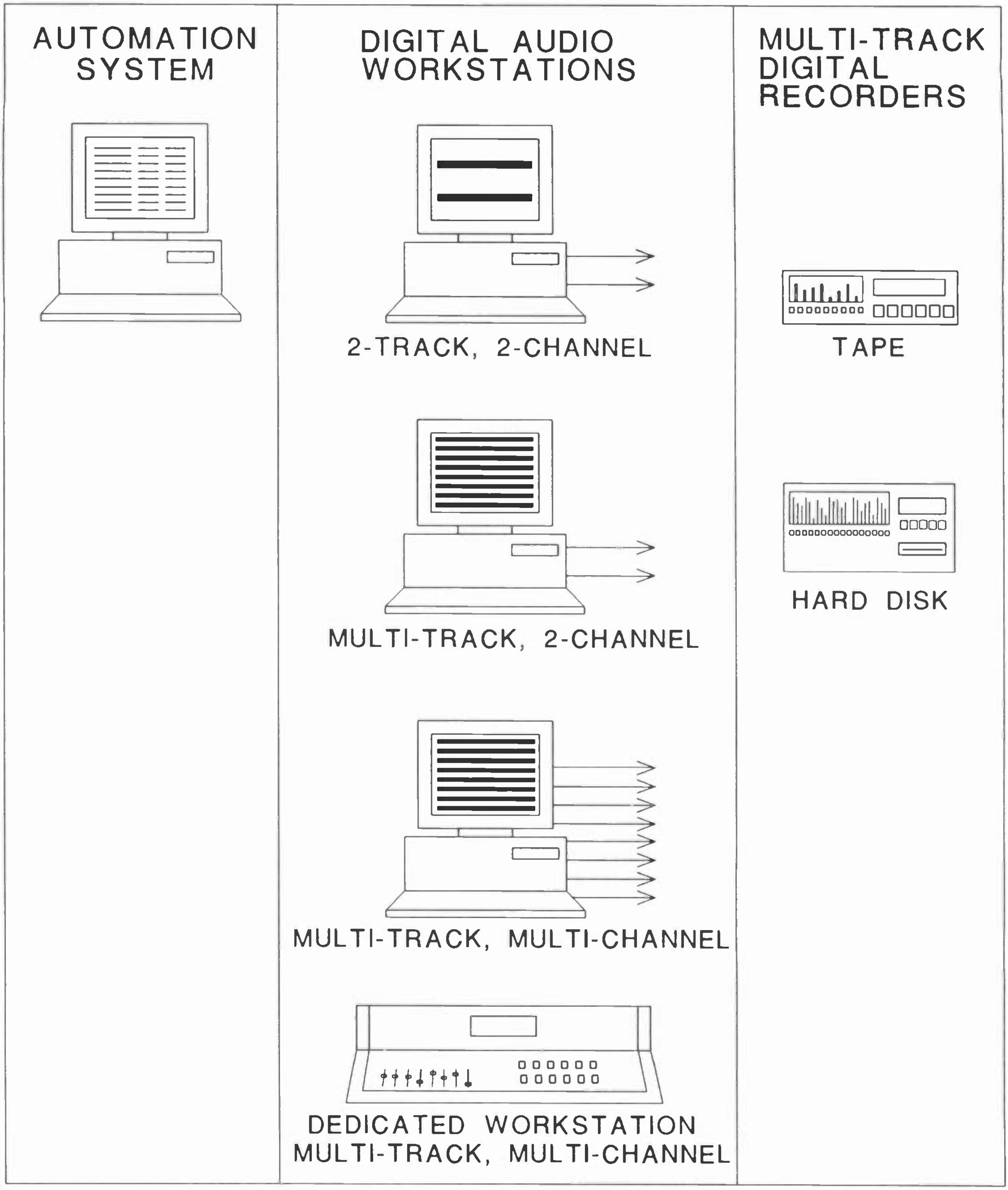
*XLR audio connectors. They cost more than RCA connectors, but interface directly with pro broadcast equipment.

*Sync to SMPTE and MIDI. This is necessary only if you incorporate video or MIDI equipment in your productions.

*AES/EBU and IEC-958 digital I/O ports. AES is the pro format; IEC is consumer. Not all manufacturers follow the standards, so ask whether the DAW is compatible with your DAT.

*High quality A/D, D/A converters. The current standard for A/D is dual 1-bit delta-sigma converters, 16 bits out, with 64X oversampling. The standard for D/A is dual 18-bit converters with 8X oversampling.

SORTING OUT DIGITAL AUDIO SYSTEMS



SUGGESTIONS FOR CHOOSING AND USING DIGITAL AUDIO WORKSTATIONS

Ty Ford
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A couple of years ago the DAWs became large enough in number, and interesting enough to be looked at. Due to the soft economy, the way most facilities' capital budgets work, and the knowledge that new technology always gets cheaper and better, many facilities wisely waited to buy.

Many are now positioned to buy. The money is in the budget and there's a lot of money at stake here. Unfortunately, the number of systems on the market calling themselves digital audio workstations has doubled from 24 to nearly 50 in the last 18 months. Sifting the facts from the fiction has become increasingly difficult. Hopefully, the information in this paper will help you make the best buy for your facility.

I don't know how it is in your town, but in the Baltimore/Washington area some post-production facilities are now being driven into digital audio to save their client base. Often their clients don't know exactly what digital is all about, but the word *digital* has become important to them. I've seen work being done that took more time and money to be done in the digital domain than it would have if it had been done in analog, but the client came away positive that the end-product was better because it was digital.

I've also worked on projects that were made incredibly easy as well as better sounding with the use of a DAW. What I'm beginning to see now is analog-based houses losing accounts. The money is going to facilities with anything from a few DAT machines and a Macintosh running Didgidesign's Pro-Tools to any one of the larger more expensive systems. The phones in the analog houses just don't ring as often any more..

DAWs In Radio, Television, Post-Production and Commercial Audio Facilities

From what I've seen so far, radio, TV and commercial facilities all have slightly different agendas.

A radio station's biggest problem is getting its' money's worth out of a workstation. Like it or not, a radio station can only take advantage of the technology if it has the people who know how to use it. At radio stations most of the production work is done by the airstaff and the production director. Things may be different at your station, but at most stations you can't get the airstaff anywhere near a multi-track tape machine, much less a digital audio workstation. That leaves the production director, the morning show producer (if you have one) and the chief engineer.

No manager I know wants to explain a $30K workstation that's just become a combination paper weight and room heater because the production director just got hired away and the contract engineer won't be in til next month. In case I'm being too vague here, what I'm saying is operating with a contract engineer and minimum-wage staff people is courting disaster.

From a management perspective, here's what you need to know. You need at least <u>two people</u> who know the system very well, usually the CE and the production director. It's important to recognize the ramifications of placing this responsibility. I've seen situations where the CE has purposely held back information from other engineers and production and programming people as a power move. If you are a general manager you need to make the call as to whether or not the people you put in charge of this sort of device can handle distribution of its knowledge without political encumbrances. That means considering what you already know about them, talking to them about how they envision this kind of system being used at your facility, and talking to the people below them in the chain to see if there's a problem with the CE or production director that's not making its way back up the chain.

For commercial audio, TV and video post-production facilities the problems are a little different. You can expect all of the engineers in a commercial audio facility to be up to speed on most of these systems in a few weeks because that's what these people do all day. Likewise, dedicated audio people in a TV or post-production facility, should be up to speed as quickly. But being up to speed with the system and knowing where the pot holes are are two different things. A post production facility in DC found out by accident that their system only stored 200 edit cues when they entered number 201. The version of the story I heard was that entering 201 blew out all of the first 200, and that they had to be re-entered.

If your video post-production facility really doesn't do much audio, you may run into the same problems that radio stations have of not having enough people who are really familiar with the system. Don't expect someone who is a whiz on a video edit controller to be able to transfer their skills to the audio workstation. Some of them will. But being intimate and seamless with a system requires that you work on an ongoing basis with the system.

CONSIDERATIONS FOR ACQUIRING A DAW

Practical Problems:
Learning curve of any computer system added to the learning curve of the software and hardware itself. (e.g. Dyaxis and Pacific Recorders requires Macintosh knowledge). Roland's DM-80, AMS Audio File, and the AKG DSE-7000 don't require fancy file maintenance. If you're expectations are that the system should be used by as much of the staff as possible, and SMPTE capability is not an issue, I suggest you look at the DSE-7000. It is arguably the simplest eight-track workstation on the market at the moment.
The minute you have more than one person working on the system, the possibility of problems increases. Unlike a production console which has switches and lights to show you its configuration, most workstations have theirs out of sight on an electronic page somewhere. Changing a sample rate for your project may not be noticed on the next job until it causes problems.

Multiple users also need to be aware of each other's files and productions. A simple housekeeping system must be set up so that you don't accidentally trash someone else's files. In addition, someone must be empowered to maintain that system. This takes time and knowledge.

I'd like to see more manufacturers develop a two-tier user interface which allows ease of use for the entry-level user, plus expert levels for more complicated work. Both the hardware interface and the software have to be designed for easy operation by entry level staff people. This system should give you everything you need to do the job now being done by a mic, cart machines, CD players and tape recorders, as simply as possible. Entry level users would need a password to open files that would allow them more powerful (and dangerous) functions.
A well thought out HELP FILE programmed to respond whenever the wrong buttons are pushed is a must. The file should be capable of telling you what you did wrong and advising you as to what your choices are. Some sort of protection should be created to keep entry level people from getting in too deep or accidentally damaging the system.

I/O flexibility: Check the number of analog and digital I/Os. Do you need to record more than two channels at a time? If so, how many?

Are all of the analog and digital outputs hot all the time or are they switchable?

Storage: Watch out for problems with thermal recalibration on large third-party HDs. Removable M/O hard drives would seem to be the answer for the moment although, for the moment, they're slower than regular hard drives. How many extra hard drives can you add? Can one or more of them be designated as a library in which to keep regularly used elements? How big can it be? If a system is advertised to contain 720 track minutes, can you record all of that on one track, or will you have to switch tracks because of system limitations? First ask yourself how long you really need to be in continuous record.

-Archiving and Backing Up: How much do you really need? Find the point at which caution and flexibility become anally retentive and back up one square. Remember to ask how long it takes to backup and restore the system. Be prepared to eat that time as down time.

-Placement in studio: The main operating controls should be facing the sweet spot of the monitors. This will require reconfiguration in some studios. If center placement of the controls is impractical, I strongly suggest installing a second set of audio monitors aimed at the workstation control area. Experiments with "making do" by not centering the workstation controls resulted in compromised hearing and neck aches. Also be aware that the RF emitted from CRTs used by workstations for display can get into mic circuits. Dynamic mics are more susceptible than most condensers mic I've tried. Positioning the workstation in the same room with open mics is not a good idea, since noise from the system's hard drive and fan create too much noise. Like people, computers like to be kept in a cool, dry dust free place. Make sure you know what extension cables, if any, may be used to distance the CPU from the controls.

-Protection: Consider an uninterruptable power source, a high-grade R.F. hum and spike protection and anti-static grounding.

-Commitment to service: Can the company swap boards over night? 24 Hour service? What do you do about your 3PM, 4PM and 5PM sessions when your system crashes at 2PM?

WHEN ATTENDING A DEMO:

-Don't Assume Anything: For example, the Korg SoundLink appeared at the AES show in NY a few years ago. It appeared to be a very robust, well-priced workstation until it was found that it wouldn't handle sections of audio any smaller than a second. The problem has since been corrected.

Know that many workstations aren't instant start because they don't have the RAM buffer to push the audio out of the machine before the read head gets to the disk.

-Take comments you hear about a competitor's system with a grain of salt: Most people don't really have time to stay up with the other forty-seven systems out there. Even if someone claims to know because he or she used to work there or because they saw the other demo themselves, BE WARY. There is no substitute for finding out yourself.

-Become familiar with the term "work around": In an effort to remain competitive, manufacturers often list identical features. Closer inspection may reveal that in order to do the same job, one system might take a lot more button pushing or a lot longer to process.

-Edits: How many edits can be played simultaneously during multi-track playback? This is usually a function of the number of read heads and the size of the RAM buffer. If you've got eight tracks of audio, and happen to have an edit on every track at the same time, you're dealing with sixteen tracks because of the cross-fades. This kind of problem never shows up in demo sessions.

-Simultaneous Record: How many channels can the system record simultaneously. Can simultaneous recording take place through both analog and digital ports?

-SMPTE Lock: Most of these systems will chase and lock to SMPTE precisely enough for mix to pix work, but if you're planning to lock the work station to another system to increase the number of tracks for an audio project be careful. We found with the the Roland DM-80 that we couldn't get a good SMPTE lock between a Studer 24 track and the DM-80. Roland was supposed to have a new box to correct the problem by now.

-Speed of System: Measure load in, edit, processing and archiving time. Usually the less expensive systems take longer time.

-Solutions for problems that don't exist: This happens a lot with computer-assisted workstations. The design engineers apply some function of the computer and try to sell it as a benefit or feature. Much of it is ear or eye candy. (e.g. wave form editing). It's nice to see waveforms, but if it takes time away from getting the job done, it's not a benefit. Rather than giving great graphics, because the computer screen can show them in really neat colors, the designers should be working on editing software that is so glitchless that you don't have to go down to the sample level to make the edit work.

-Automation: While it's essential for long form productions where you'd have to go back and remix 10 minutes if you blew one move, but it's eye candy for spot work. If you've got so much going on in a 30 or 60 second spot that you can't control eight tracks in real time, your creative department is out of control.

-EQ and reverb/delay effects: I see two schools of thought. The record "dry" and then sweeten on mix crowd (who normally come from the music recording part of the spectrum); and the "build-it-as-you do-it" crowd who come from radio stations and project studios. The music recording people do it the way they do because when you've got 16, 24, 32 or 48 tracks of audio, it's literally impossible to tell how much sweetening is needed to make the mix work. However when you're doing four or eight track work for spots, promos and long-form productions, you can make most of those decisions by listening as you lay in the tracks.

Listen carefully to the onboard digital EQ, reverb and delay of any system. The prevailing wisdom is that 16 bit digital audio, while acceptable for I/O interfaces, does not offer the definition needed for effects processing. Using lower than 20, 24 or higher bit streams amy result in unacceptably grainey or metallic sounding artifacts.

-Best Advice: Get them to do the demo at your place. Nothing takes the place of seeing how a system works in your own studio. Let them do their demo, then do yours, with them at the controls. Have as many users on hand as possible to take notes and ask questions. Once the company does their demo, have them do yours. In fact do several. One of a simpler nature for those operators who will be doing basic work, then one that's as complex as it gets to make sure the system can do the job more elegantly and in less time that your current setup.

Having covered these dim areas of caution, I can tell you that working with a digital audio workstation will change the way you work. After you consume the specifics of whatever system you choose, you'll begin to create an evolutionary path that is the result of your own experience with the system. The longer and more intimately your efforts, the more you'll be able to do. When that happens, you will begin to loathe working with tape transports and linear formats.

Although the lack of noise digital audio affords us is great to listen to, it is the power of non-destructive editing that makes it worth the effort to learn a system. The power of knowing that you can try any edit and undo it is aphrodisical.

A task like performing a 1/60 of a second edit to remove a tongue-click from the middle of a word is very satisfying. So is knowing that, by pulling the 1/30 to 3/30 of a second centers out of breaths in a read, your voice-track is perfectly timed for the donut.

If you've been frustrated by a lack of timing precision with linear formats, you'll really enjoy being able to place audio exactly where you want it. The impact of your production is increased because of the preciseness with which you can place each element.

While it's true that a good portion of the "time saving" that workstation makers speak about is eaten up in experimental editing, there are many features that do save time.

Locating to to the head, tail or other marked points is done in less than a second on most systems. The same is true for cutting, copying and pasting sections of audio. Once you get the audio loaded in, doing multiple versions of a production can be done very quickly.

Once you get into it, you'll never want to go back to tape. For the radio station people, If you liked the move from vinyl to CDs, you'll love a well-designed random access digital editor. After working with the AKG DSE-7000 for over two years, I can tell you , the pain is definitely worth the gain.

NEW TELEVISION TECHNOLOGY

Sunday, April 18, 1993

Moderator:
Louis Libin, National Broadcasting Company, New York, New York

IMPLEMENTATION OF GHOST CANCELERS IN HOME TELEVISION RECEIVERS
D. Koo, A. Miron, C. Greenberg, S. Herman, and C. Tung
Philips Laboratories
North American Philips Corporation
Briarcliff Manor, New York

***ADVANCES IN THREE DIMENSIONAL COLOR DECODING TECHNIQUES**
Martin Weston
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***ALTERNATIVE VIDEO SIGNAL MONITORING**
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Beaverton, Oregon

*Paper not available at the time of publication.

IMPLEMENTATION OF GHOST CANCELERS IN HOME TELEVISION RECEIVERS

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Abstract: The year 1992 witnessed a major milestone towards the realization of ghost cancelation for NTSC television. This was the adoption by the Advanced Television Systems Committee (ATSC) of the voluntary ghost cancelation reference (GCR) signal and the corresponding allocation by the FCC of the required vertical blanking interval (VBI) resources. That is expected to induce broadcasters to transmit the GCR, giving receiver manufacturers the incentive to build ghost cancelation abilities into their receivers. This paper gives an overview of the current status of this task, with an emphasis on the design of the in-receiver ghost canceler.

WHAT ARE GHOSTS AND HOW ARE THEY CANCELED?

TV ghosts are multiple images caused by multipath echoes. Television signals travel along straight lines. The receiving antenna may have a short direct path between it and the transmitter antenna. The same signal can reach the receiver antenna over a different, longer echo path, due to reflections off nearby buildings and other objects. Under such conditions, the receiver will show two images, a strong main image and a weaker shifted echo or "ghost". Such ghosts may seriously degrade the received image quality.

Ghosts are canceled in the receiver with a two-step process:

1. Characterize the ghosting channel
2. Cancel the ghosts using adaptive filters.

To facilitate the channel characterization step, the broadcasting station transmits a "Ghost Cancelation Reference" (GCR) signal. The receiver compares the received, ghosted version of the GCR signal with a clear, stored replica of the same signal. Such comparisons are computed by a digital signal processor chip. Then this chip uses special ghost cancellation algorithms to calculate the coefficients to be fed to the digital adaptive filters that cancel the ghosts. This two-step process occurs continuously. The coefficients of the digital filter are being constantly updated, to follow transient conditions, as measured from the received GCR. A simplified block diagram of the Philips ghost canceler system is shown in Figure 1. The received analog composite baseband video signal is converted to digital form by an A/D converter. The ghosted received GCR is captured. From it settings are computed for the ghost canceling filter. The corrected signal, with the ghosts removed, is converted back to analog for normal video processing and display.

THE GHOST CANCELATION REFERENCE (GCR) SIGNAL

The GCR signal is the key link between the broadcaster and the receiver. The Advanced Television Systems Committee (ATSC) undertook an exhaustive set of tests of the performance of five competing GCR candidates, at the request of the NAB. As a result of these tests, the ATSC voted unanimously in August of 1992 to recommend the Philips GCR as the USA standard signal. The new American GCR standard was invented by David Koo of Philips Laboratories, Briarcliff Manor, NY. While the GCR signal looks like a swept-frequency chirp pulse, as shown in Fig. 2, it is in reality a signal which has been carefully synthesized to have a flat frequency spectrum over the entire 4.2 MHz band of interest to NTSC applications. (The spectrum of a linear FM chirp pulse is not flat.) The GCR signal is broadcast after insertion in one of the vertical blanking interval (VBI) lines. The FCC gave its approval on October 27, 1992, for the substitution of the GCR into line 19 of the VBI, making the Philips GCR and its VBI location the official USA voluntary standard. The voluntary nature of the standard relies on the cooperation of the broadcasters. Most broadcasters who have been contacted were enthusiastic about the possibility of substantially improving the quality of the received images for a very low-cost investment in additional equipment at the transmitter site. In fact, field tests conducted by the ATSC have shown that ghosted images were improved

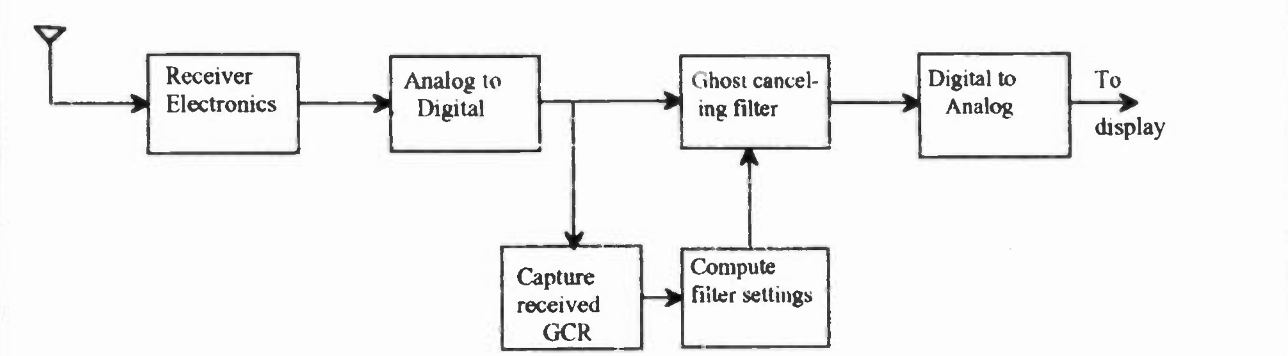

Figure 1. Simplified block diagram of a ghost cancelation system

about 95% of the time as a result of ghost cancelation. The average improvement was about 1.5 points on the 5-point CCIR scale. The magnitude of the improvements were such that most ghosted images had no perceptible ghosts after cancelation.

GHOST CANCELERS IN HOME RECEIVERS: A STATUS REPORT

The ATSC experiments outlined above proved that ghost cancelation can make a major improvement in the quality of the received TV images. The broadcaster can do his share to contribute to this improvement for the very low cost of a few hundred dollars per transmitter site. However, the broadcast GCR can only be used by receivers that contain ghost cancelers. At the time of this writing, the only ghost cancelers being sold in the USA that can use the new GCR are the VECTOR professional cancelers being manufactured by Philips Broadband Networks Inc. of Manlius, NY. Many receiver manufacturers are now working to reduce the cost and size of the ghost canceler modules so that they can be economically included in home TV receivers. A working prototype receiver with onboard ghost cancelation is being demonstrated by Philips at the NAB93 Convention. Meanwhile research and development is continuing on finding the optimum tradeoffs, the best way to reduce the cost and size of the hardware, while minimizing compromises in performance. This section will outline some of the tradeoff issues that are being considered.

- **Reduced system performance requirement:** Professional cancelers have to satisfy the very rigid quantitative performance requirements that are imposed by broadcasters and cable operators. In home receivers, however, it is sufficient if the vast majority of the ghosts are reduced to visually imperceptible level. As a consequence, for example,

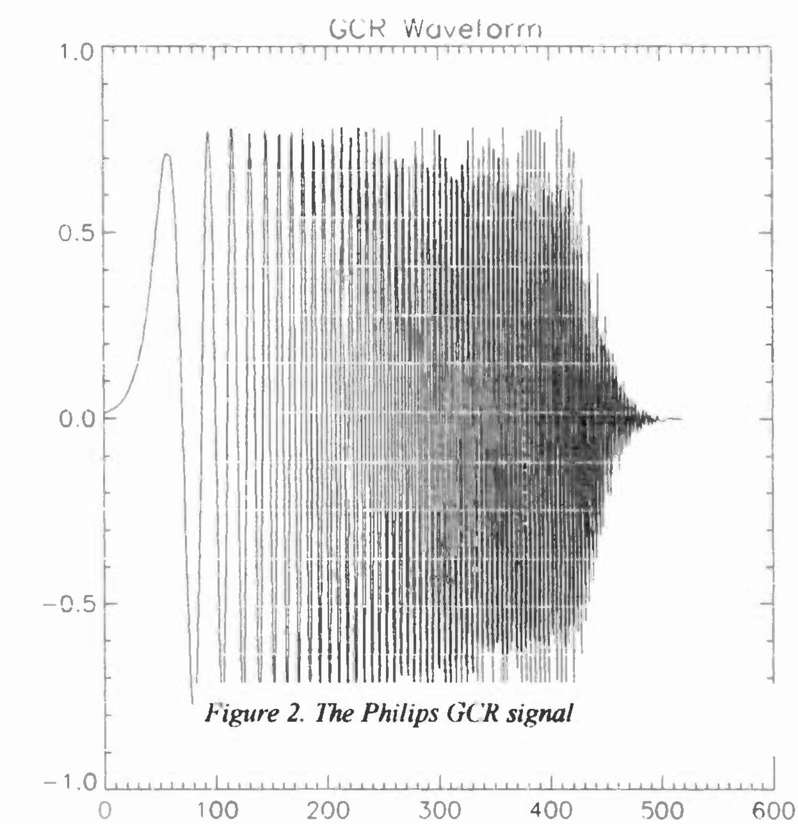

Figure 2. The Philips GCR signal

it may be possible to reduce the number of bits to which the video data is being digitized. At this time the professional unit uses 9-bit video data and 8-bit filter multiplier coefficients. It is believed that in consumer units 8-bit or, perhaps even 7-bit data may produce canceled images of very good quality, at a substantial savings in silicon costs.

- **Delay-range compromises:** The specifications for the generic professional canceler call for canceling ghosts in the range of -3μs to +45μs. In fact some cable applications call for canceling at pre-ghost ranges greater than -3μs. In special applications, there is the potential to alter the VECTOR to have a pre-echo range of up to -45μs. In consumer units, advantage may be taken of the fact that the vast majority of ghosts are contained in a substantially shorter delay range. This would help save digital memory costs.
- **The number of ghosts that are to be canceled:** The professional unit is designed to cancel up to 13 *groups or clusters* of ghosts, where each group can contain several closely spaced echoes. This is obviously an overkill in most practical situations. To achieve this, the professional unit needs two PGC180 digital filter chips for a total of 360 multiplier taps. These taps are dynamically allocated. In a typical situation, about 100 are used in the FIR filter portion to cancel pre-ghosts and perform frequency equalization. Whereas 260 multipliers are used in the IIR filter to cancel post ghosts. It is believed that in the consumer unit it will be sufficient to use only one PGC180 filter chip and allow the software to allocate the taps to the delays where they will do the maximum benefit. For example, a typical application may still have 100 taps for the pre-ghosts and equalization. However, 80 taps may often be satisfactory since they can cancel up to 4 clusters of ghosts. In any case the filter chips and the associated systems have been designed to continuously reallocate the utilization of available multiplier taps to be used in the most efficient manner.
- **Choice of the computer chip:** A computer or digital signal processor (DSP) chip is needed to perform the necessary calculations and control the ghost cancelation process. In the professional unit a Texas Instruments TMS320C25 chip is being used. In consumer units it may be possible to substitute a cheaper microcontroller, with a slight compromise in cancelation speed.
- **Need for coefficient storage:** Professional units are normally used while being tuned to a single channel for long periods of time. On the other hand, consumers often do rapid "channel hopping". To accommodate to the nature of the consumer usage, additional digital memory is allocated in the consumer version to store initial approximations to the filter coefficients for each channel.
- **Lower digital chip voltage:** It may be decided that the supply voltages of the digital chips will be reduced from the levels now being used in the commercial units. This would greatly reduce the cooling requirements and still operate the chips within design specs.
- **Lower-cost silicon:** Initial prototype ghost cancelers were built with off-the-shelf components. However such systems are much too expensive for the consumer market. Economic solutions demand custom-designed components. Using such custom components, there is every reason to believe that the chips being used for ghost cancelation will undergo the same cost reduction cycles as have been experienced in other fields of electronics. Indeed, the PGC-180 chip is Philips' second generation ghost canceler filter chip. It is anticipated that future generations of chips will cost substantially less and incorporate more functionality, such as other TV control/processing functions.
- **Choice of algorithms:** Various algorithms are used to compute the characterization of the ghosting channel and the digital filter coefficients. The choice of algorithm affects tradeoffs between the costs of the computer chip, the speed of cancelation, the number of ghosts that are canceled and the amount of residual ghosts (if any).
- **Ghost canceler options:** The goal is to eventually provide for the consumer a range of ghost cancelation options. These will be matched to the application. For example, high-end sets with high quality displays will require more complete ghost cancelation than low-end sets with small screens of low resolution. In addition, it will be desirable to provide for set-top cancelers to allow consumers to benefit from ghost cancelation without the need to purchase new receivers.

SUMMARY

The era of practical ghost cancelation in the USA is now here, with the adoption of a voluntary American GCR standard signal. Ghost cancelation has been shown to be able to substantially improve the performance of home TV receivers. While the process of implementing the GCR standard requires that the broadcasters transmit the GCR signal, it is also essential that the manufacturers produce receivers that use this GCR to complete the cancelation process. The broadcasters can perform their share of this procedure at a very low cost and very rapidly. Indeed, many are already doing so. However, there is a much longer latency period before substantial numbers of home receivers are sold with built in cancelers. How fast this occurs is, to a large measure, a function of how much the cost of the devices can be reduced without seriously compromising its performance. This paper gave the perspective of one TV receiver manufacturer on the options that exist. This perspective has been aided by the experience gained in the research, design and development that led to the manufacture of the first and only professional canceler that is available at the time of this writing with the ability to use the new GCR standard. With that canceler as a starting point, a set of possible compromises, and the consequences of each compromise, were outlined. The design process is still proceeding and it is not possible at this point to predict the exact configuration that actual consumer units will emerge with. However the current goal is that those first units be substantially cheaper than the professional unit, and that they satisfy the ghost-cancelation needs of the home viewer with high-quality performance.

AUTOMATED TELEVISION REMOTE CHECK-IN SYSTEM

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ABSTRACT

A number of factors complicate the check-in process commonly performed prior to a remote broadcast. In the past it was performed manually, making it a time-consuming and personnel-intensive task. This paper describes how the benefits of automation can be applied to the remote check-in process using off-the-shelf equipment.

The Technical Challenge

Remote origination for news, sports and entertainment programming has always presented technical challenges. For example, in the past four years CBS has recorded a total of 91 transmission-related problems on sporting events alone. Efforts to minimize transmission problems commonly include a remote check-in, which is an evaluation of common carrier link fitness prior to air time.

Several issues complicate the remote check-in process. Common carrier transmission lines typically come up only two hours before a broadcast begins, so if problems are uncovered in the check-in process there is very little time to correct them. Also, the remote production crew often needs access to the lines before air time to do pre-feeds, thereby reducing the time available for check-in even further. Audio and video transmission paths are becoming more complex, possibly involving multiple carriers with varying services including microwave, fiber, telco, and satellite links (C and Ku band).

These technical factors, coupled with management's mandate for reductions in operating costs, increase the demands placed on technical resources supporting such broadcasts. Manual check-in procedures no longer satisfy the time constraints or technical needs of CBS.

Developing a Solution

Recognizing these challenges, CBS began development of a system to improve the overall quality and efficiency of remote broadcast productions. We determined that automating the remote truck check-in process was the key to unlocking the solution.

CBS began its efforts three years ago with a series of experiments to test, and hopefully demonstrate, the feasibility of an automated remote check-in system. The first experiments verified only common carrier audio circuits using a remotely controlled audio generator and modem interface.

These early tests made it clear that an automated remote check-in system of this type would pay for itself if it could provide the following advantages over the manual check-in process:

- reduce check-in time
- reduce manpower requirements in TX
- eliminate field personnel from process
- free up field truck audio console and video switcher
- improve signal quality through more comprehensive verification of transmission path performance
- automate transmission path measurements
- reduce common carrier line costs

The initial attempt at developing a complete audio and video check-in system was predicated on the use of audio and video signal generators with remote control capability. The plan was to control these units via a custom controller, which would be programmed to call up the desired test signal at the remote site. A time lock would be incorporated so that during program time the signal generators would be locked out to eliminate the possibility of program interruption.

This system could not provide automated measurement of the received signal, and was therefore unable to deliver all the desired benefits listed above. However, it did represent a significant improvement over the existing manual check-in procedure. As a result of changing priorities and workloads, this proposed system was never implemented.

From the time the original system was conceived, almost two years passed with little progress on the system. During that time, however, products that could satisfy all the major system objectives came to market. Finally, a collaborative effort between CBS and Tektronix culminated in the development of a new system that employs only off-the-shelf components and satisfies all major performance objectives.

System Overview

The latest system is based on a Tektronix VM700A Video Measurement Set with Option 40 Audio Measurements (referred to as "measurement set" for the rest of this paper). It is located in the CBS transmission area (TX) and communicates with Tektronix ASG100 Audio Signal Generators (audio generators) and Tektronix VITS200 NTSC VITS Inserters (VITS inserters) in a CBS remote truck. Communication is through a pair of secure modems and a code-operated switch, all of which are stock items from the Black Box Company. Figure 1 shows a simplified block diagram of the system.

Through automation, the measurement set conducts a thorough set of tests in a fraction of the 20 to 25 minutes previously required for this process. The manual video test procedure typically employed color bars and a character generator to check video circuits for levels and intercarrier interference. Manual audio testing utilized 50, 400, and 10,000 Hz test tones to test for frequency response, headroom, and signal to noise ratio. The new, automated procedure is far more comprehensive, performing a broad range of video tests from the measurement set's repertoire and thorough series of audio tests based largely on the multitone signal and tone sequences.

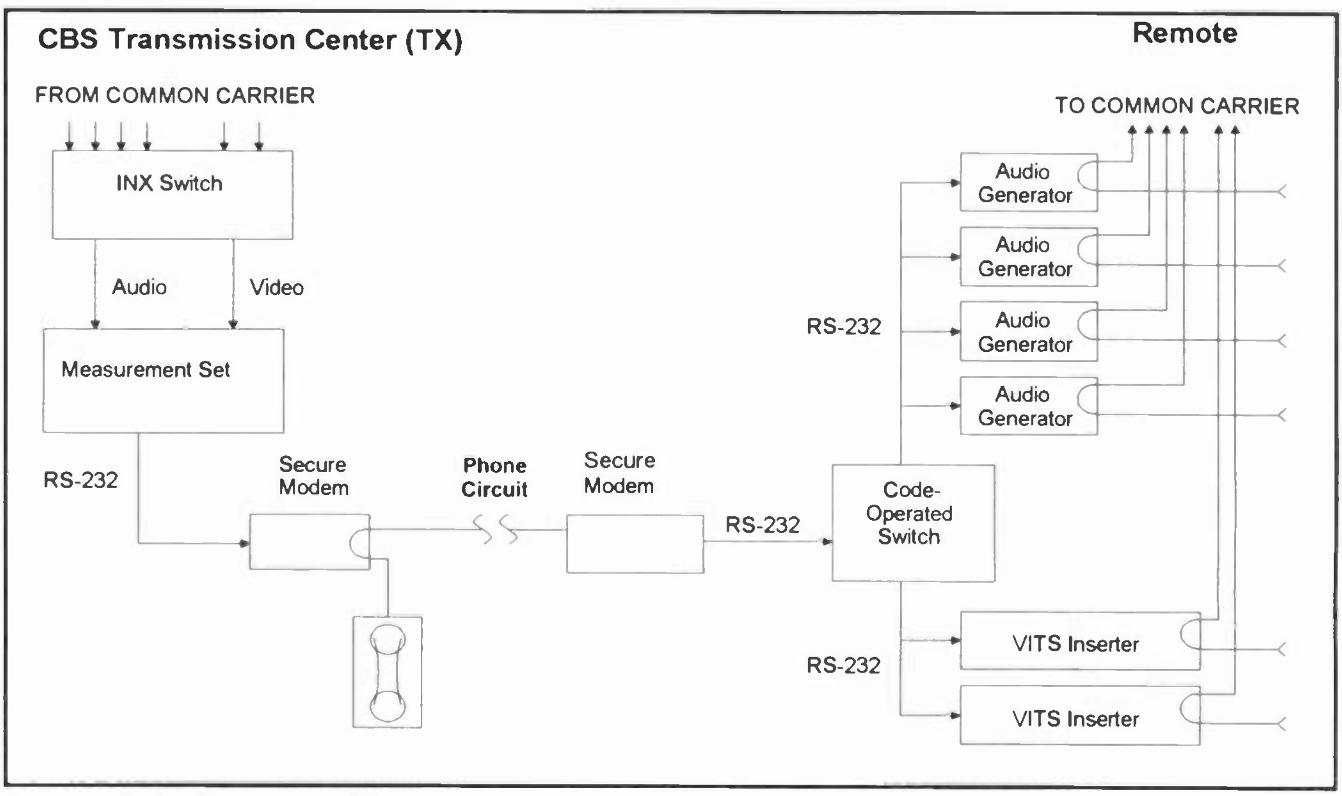

Figure 1. Block Diagram of CBS' automated remote check-in system.

CBS Transmission Center

The following system components are located at TX in New York:

- incoming signal switcher test bus (existing bus of CBS INX switcher)
- secure modem
- measurement set
- ancillary audio and picture monitors

Figure 2 outlines the equipment configuration in TX. Initially, the system was configured for four audio (stereo or mono) and two video circuits, but it could be sized to suit almost any requirement.

The TX operator controls the check-in system with two devices; the measurement set and an input switcher (INX). The measurement set uses preprogrammed functions to control all the remote equipment and automatically measure the incoming audio and video signals. (Functions are user-created macros that reside on the measurement set. Functions can control external equipment, execute lists of measurement set applications, display screen prompts and pause the instrument. Functions can be nested several layers deep, i.e., one function can call other functions that can, in turn, call yet other functions. A touch-screen interface makes starting functions a simple matter of touching the exact function name displayed.) To route the appropriate signals to the measurement set, the operator controls the INX switcher through a keyboard interface. With this setup, one operator in TX can perform a complete remote site check-in single-handedly.

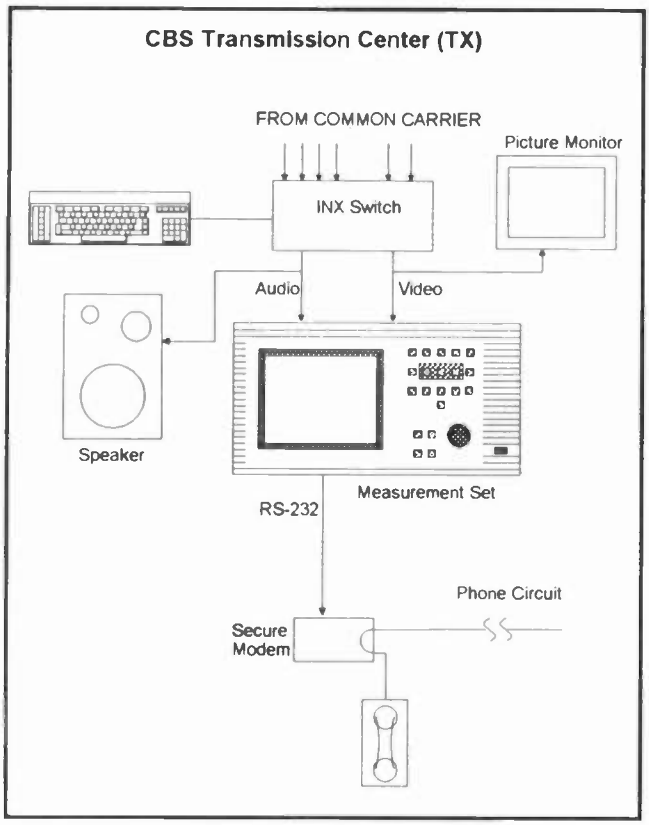

Figure 2. Diagram of monitoring and control equipment in TX.

Remote Truck

The remote truck contains these system components:

- secure modem
- RS-232 code-operated switch
- four audio generators
- two VITS inserters
- test/normal selector and technical director control/indicator panel

Figure 3 outlines the remote equipment configuration. Once a phone link is established, the secure modem at the remote site receives control commands issued by the measurement set and passes them to the code-operated switch. Access codes select the appropriate switch output for subsequent commands.

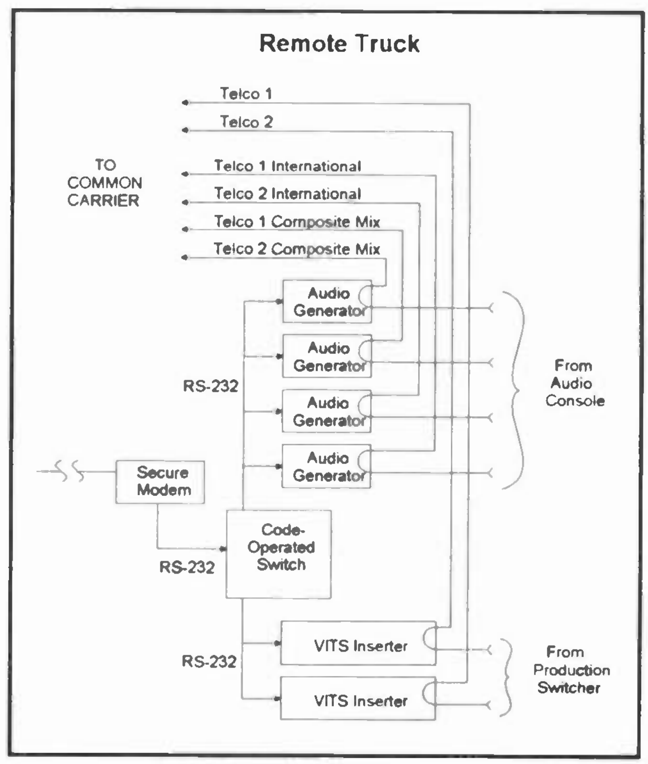

Figure 3. Diagram of equipment in remote truck.

The audio generators are in-line with the four audio circuits: Telco 1 Composite Mix (event and commentary), Telco 2 Composite Mix (backup), Telco 1 International (event sound only), and Telco 2 International (backup). The two video circuits, Telco 1 and Telco 2 (backup) pass through the VITS inserters.

During the automated check-in all these devices are controlled exclusively by commands received from TX. The truck maintenance supervisor, audio operator and field technical director are free from any involvement in the check-in process, as are the truck's video switcher and audio console.

Communication System

Security

Security is very important any time a phone line controls equipment that could take a program off the air. The secure modems chosen provide a very high level of confidence that programming will not be interrupted by someone accidentally or intentionally connecting to the remote modem. A hardware bypass for the audio generators and VITS inserters adds an additional layer of security.

Dial-up Process

Because one of the goals of this system is to eliminate the involvement of personnel at the remote site during check-in, an operator at TX initiates the phone link with the mobile unit.

The first step in the process is to run a function on the measurement set. This function puts the measurement set into a loop that continually checks its RS-232 control port for carrier detect.

Then, using a handset, the operator manually dials the number of the mobile unit. Since the mobile unit could be anywhere on the continent, having the measurement set dial the mobile unit would require modifying functions in the measurement set every time the mobile unit changes locations.

When the operator hears a tone from the remote modem, the next step is to flip a switch on the modem in TX to the data position and then hang up the phone. Once the two modems lock and the carrier detect line is raised, the function running on the measurement set notes the carrier detect status, stops looping, and issues a password to the secure modem at the remote site.

Upon receiving a *valid* password, the remote modem immediately disconnects. It then searches a look-up table for the call-back number associated with that password, and redials that number (the secure modem at TX). This procedure prevents anyone from intentionally or accidentally connecting with the remote modem and taking a program off the air. If the modem receives an invalid password, it simply disconnects and awaits the next call.

After the remote modem receives a valid password and disconnects, the measurement set drops into a second loop, once again waiting for carrier detect to go high. When the remote modem calls back and connects with the modem at TX, the measurement set displays the message "CONNECTED." Touching the screen clears the message and displays four directories containing the functions that perform the actual check-in tests.

Code-operated Switch

At the remote site, the secure modem is connected to an RS-232 code-operated switch. The switch receives commands from the measurement set through the phone link/modems, and passes them on as directed to the four audio generators and two VITS inserters.

To select any of the six instruments, measurement set functions send the appropriate control characters to the switch. For example, "^D 1" selects the first of the four audio generators, and "^D 2" selects audio generator 2. The selected device receives all commands that follow until a new string of control characters directs the switch's output to another device.

Measurement Set Control Functions

A total of 63 functions were written specifically for this remote check-in system. Pressing the measurement set's front panel "Function" key displays four directories that contain all the custom functions. Figure 4 maps the basic directory structure.

- ALL_TESTS contains functions that initiate all the audio and video tests routinely run during a remote check-in.
- AUDIO_CONTROL stores functions that actually perform the audio tests initiated by functions residing in the ALL_TESTS directory.

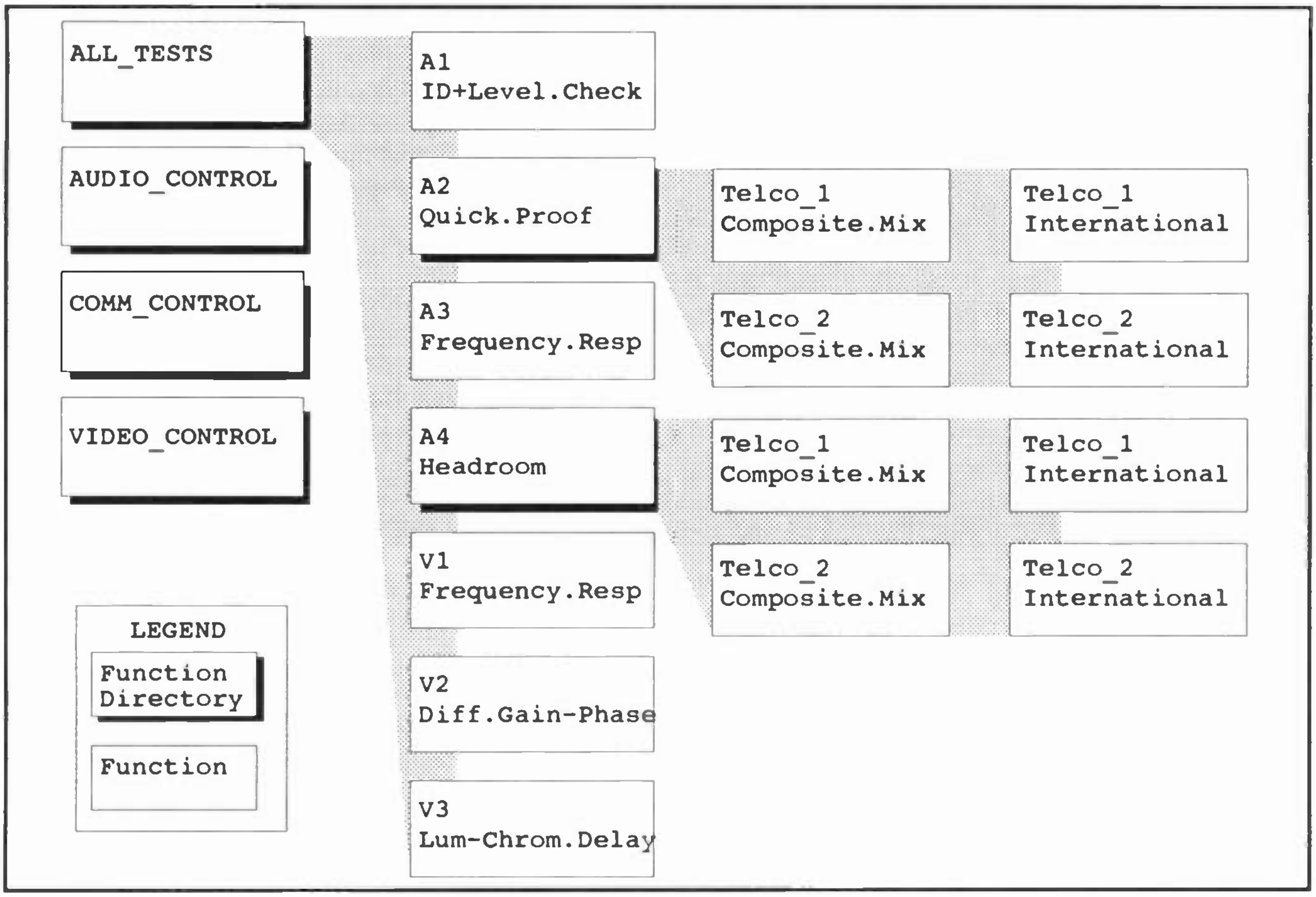

Figure 4. Structure of main function directories and files for the remote check-in system.

Functions to select and control each audio generator for troubleshooting purposes are also located in this directory, in addition to all the screen prompts and pass/fail messages for audio tests.

- COMM_CONTROL holds functions that establish the phone connection between the measurement set and the remote site, and functions to restore the program path in the mobile unit and break the phone connection after the check-in is completed.
- VIDEO_CONTROL contains functions to perform the video tests found in the ALL_TESTS directory. Additional functions to select and control each VITS inserter for troubleshooting purposes are also located in this directory.

Standard Tests

From the main function screen, touching the ALL_TESTS directory reveals five functions and two additional directories. The operator at TX simply works his or her way down through the list, verifying that each audio and video circuit passes all tests.

Notice the A1, A2, A3, A4 and V1, V2 prefixes on each of these functions (Figure 4). These prefixes cause the functions to appear on the measurement set's display in the order of normal execution.

The following sections describe how the custom functions operate and the interactions required by the operator at TX.

ID and Level Check

Identification of incoming audio and video feeds is of primary importance, so the first test function run from TX automates this process. The function "A1 ID+Level.Check" starts by calling functions that instruct both VITS inserters to output color bars with ID text (e.g., CBS MU11 Telco 1).

Next, the function instructs each of the audio generators to output a 400 Hz, 0 dB tone that alternates with a voice ID (e.g., "This is CBS MU11

Telco 1 Composite Mix") in a continuous loop. At the same time, the prompt VERIFY AUDIO ASSIGNMENTS & LEVELS appears on the measurement set's display.

The TX operator routes the two video sources, one at a time, to a picture monitor to verify that both feeds are on the correct inputs of the INX switcher.

The TX operator then selects the audio feeds, one at a time, and checks the measurement set (which has dropped into the audio monitor mode) for proper levels. Loudspeakers in TX permit identification of each audio feed via the voice ID. In the case of stereo feeds, the audio monitor display also indicates polarity reversal.

When the circuit ID process is complete the measurement set drops back into the function mode and the system is ready for the next phase of the remote check-in.

Audio Quick-Proof

Audio quick-proof functions provide at-a-glance verification that left and right channels are not swapped, XLR polarity is correct, and that there are no serious frequency response or distortion problems. Separate quick-proofs functions for each audio circuit simplify the procedure (Figure 4).

Touching the quick-proof function "Telco_1 Composite.Mix" starts the audio auto-test on the measurement set and prompts the TX operator to route that circuit from the INX switcher to the measurement set. The function then selects the TEK:91 tone sequence on the appropriate audio generator and puts the generator on line.

Results appear a few seconds later on the measurement set. Figure 5 shows a typical display from this function. If all is well, touching the screen brings back the list of quick-proof functions so the next circuit can be exercised. If errors are detected, the TX operator can troubleshoot the problem with other functions available on the measurement set.

Audio Frequency Response

Once audio feeds are identified and given a quick check, audio frequency response is evaluated in greater detail. The function "A2 Frequency.Resp" instructs all four audio signal generators to output a multitone signal. It then starts the measurement set's multitone application and displays the prompt ROUTE EACH AUDIO SIGNAL TO MEASUREMENT SET TO VERIFY FREQUENCY RESPONSE.

The TX operator cycles through the four audio feeds and verifies frequency response of each at a glance.

Audio Headroom

Before the automated system was developed, verifying audio headroom through the common carrier link was by far the most troublesome task in the check-in process for the crews at both ends of the broadcast. It was also the check most likely to fail. The operator at TX had to be on the phone with the mobile unit throughout the process and, if any problems were encountered, phone a contact at the common carrier.

Now, with the automated system, the operator at TX simply touches a function key and waits a few seconds for a pass/fail message, and repeats the process for each audio feed. If any common carrier channel fails to provide adequate headroom, the

```
Audio Measurements

At            Fri Jan 15 14:58:51 1993   Video Source: A
Test Type     Tektronix Program 91   Expected TEST level: 0 dBu
Source        Telco 1 Composite Mix
                                                                  Violated Limits
                                        Left         Right          Lower   Upper
Insertion Gain Error (dB)               0.00         0.05
Sweep Max. Gain (dB)                    0.10         0.06
Sweep Min. Gain (dB)                   -0.09        -0.04
THD+N (at 400Hz) (%)                    0.034        0.022
Polarity                              Normal       Normal
Stereo Channel Assignment             Normal       Normal
Crosstalk (into channel) (dB)         -68.02       -67.07
ANSI SNR (weighted) (dB)               87.53        86.81

Gain Difference (dB)                   -0.05
Phase Difference (deg.)                -0.01
```

Figure 5. Typical display from the measurement set's audio auto-mode.

operator at TX can control the remote audio signal generators via functions on the measurement set for troubleshooting purposes. The common carrier must still be contacted by phone if a problem exists, but at no time does the headroom check involve anyone from the remote crew or tie up any of the operational equipment.

Simplifying the task to this extent required very clever use of nearly every feature available in the measurement set's functions. The interactions between the functions involved are quite complex, as this brief overview of one of the four channels suggests.

The displayed function "Telco_1 Composite.Mix" (Figure 4) starts the process by selecting the appropriate audio generator and starting the measurement set's audio analyzer application to measure THD+N. Two subfunctions do most of the work.

"Step_Level" establishes a 10 kHz, 0.0 dBu output, and calls the function "Check." Check determines whether or not a predetermined distortion level has been exceeded. If it has, an error message is displayed on the measurement set. If not, Check returns control to Step_Level. Step_Level increases the tone level by 1 dBu, and calls Check again. This loop of increasing tone level and checking THD+N continues until an audio level high enough to ensure adequate headroom through the common carrier link is met or until a distortion level above the threshold stops the process. In either case, the TX operator gets an on-screen message reporting the pass/fail status of that channel.

Checking the three remaining channels is done in the same manner, i.e., by selecting the function bearing the name of the audio circuit and routing that audio circuit to the measurement set.

Video Frequency Response

The function "V1 Frequency.Resp" signals both VITS inserters to output FCC multiburst and then starts the graphical multiburst measurement mode. The TX operator then switches between the two video sources and checks the display for adequate response.

Differential Gain and Phase

Like video frequency response, the function "V2 Diff.Gain-Phase" sets the VITS inserters' outputs for the appropriate signal (NTC7 Composite in this case) and initiates the proper mode on the measurement set (Diff Gain/Diff Phase mode). The TX operator switches between the two sources and checks that both fall within preset limits.

Chrominance-to-Luminance Delay

The function "V3 Lum-Chrom.Delay" sets the VITS inserters' outputs for NTC7 Composite and initiates the Y/C delay mode on the measurement set. Again, the TX operator switches between the two sources and checks that both fall within the preset limits clearly marked on the display.

Troubleshooting

If any of the standard tests in the ALL_TESTS directory fail, the measurement set at TX is equipped with functions to select and control the remote audio and video generators. The interface is such that routine problems can be easily handled by an unskilled operator without the help of a remote crew.

By using functions in the AUDIO_CONTROL directory, the operator at TX can select the audio generator on any of the four circuits. Practically speaking, the bulk of the audio generator's functionality has been packed into a handful of functions stored in this directory. These give the operator the power needed to sort out weaknesses in the link. The functions included can select the different tone sequences, a line-up tone, several discrete tone frequencies, different multitone signals, voice ID and silence.

The VIDEO_CONTROL directory gives the TX operator the same capabilities for troubleshooting the video link as supplied for the audio link. On either VITS inserter, Color bars, FCC multiburst, NTC7 composite and combination, and (sinx)/x signals can be selected easily.

Conclusion

The automated check-in system CBS and Tektronix have developed delivers significant manpower and cost savings. Automating the check-in process frees the remote facility from the burden of check-in support and improves control over (and confidence in) remote broadcast originations. In-service testing of video lines is also possible.

Certification of transmission line performance may now be done quickly and easily by one operator in the CBS Broadcast Center completely independent of the remote truck. The rapid handling of problems in most cases gives production earlier access to transmission facilities.

All this has been accomplished with standard, off-the-shelf hardware customized through the programmable functions within the measurement set. The system is fast and adapts easily to changing requirements.

Although it has not yet been explored, common carrier access to this test system could be permitted prior to the scheduled availability of service. This would allow the carrier to completely check out transmission lines and guarantee performance well in advance of the scheduled availability. Broadcasters would then be assured that service would commence on time and with minimum risk of technical problems.

This technology lends itself to many settings where operators control remote equipment. Future applications will very likely be found within large broadcast plants, common carrier facilities, and even within stand-alone studios when facility performance is being tested.

ON-LINE SPOT BUFFERING SYSTEM VIABILITY STUDY

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This paper describes a commercial spot buffering concept to improve the efficiency of multicassette spot playback systems, and gives results of a detailed study of spot traffic and on air usage which provide some solid numbers about peak library size, and daily new spot filings into the system. Results of a computer simulation of the buffer system prove the concept could provide a four-fold increase in commercial output from one multicassette machine. By using relatively inexpensive spot buffers, one cart system can be upgraded to generate multiple commercial streams.

INTRODUCTION

Television networks typically originate a television program with a different mix of commercials for different parts of the country, for example sending a commercial for snow tires to New England, and at the same time sending a commercial for all-season tires to California. Since all the regionalized commercials occur at the same time, a commercial videotape player is required for each separate commercial stream. This results in low utilization of the expensive commercial playback equipment, and errors in moving the inventory of commercials between playback equipment.

INEFFICIENCY OF CURRENT SYSTEM

CBS uses eight Sony Library Management Systems (LMS) for spot playback to the New York area, national networks, and syndicated program services. Since the regional commercials for a common program all play at the same time, this ties up a large number of cart machines.

The same problem occurs for any facility involved in program origination for multiple channels. Either you must dedicate cart machines to each program channel, or make up commercial spot reels in advance.

The problem is there are just not enough commercials on TV. Multi-cassette commercial cart machines cost a factor of five or ten what program playback equipment costs, yet these expensive machines sit idle more than eighty percent of the broadcast day.

Government regulations limit the maximum commercial time per hour for various types of programming. All four commercial networks program below these maximums, approximately ten minutes per hour. This means your cart machine sits idle 83 percent of the time. Imagine if you visited a General Motors factory and saw their assembly line shut down for fifty minutes each hour ?

Another drawback of the current approach is the need to dub multiple copies of spots and transport them around between the cart machines. Originating several channels of programming means that a number of channels may require the same spot at different (or the same) times during the day. This leads to many manual tape handling tasks, such as searching for a missing tape, purging of stale-dated spots, and last minute dubs for multiple copies.

The first step away from these problems is upgrading the cart machine to support multiple spots per cassette. This will increase by ten to twenty times the on-line spot capacity of the cart machine. Going to a multiple spot per tape format will eliminate the manual tape handling requirements, and tape management functions will be maintained internally by the cart machine.

The second step toward solving these problems is to decouple the cart machine from direct to air play.

SPOT BUFFER SYSTEM

The spot buffer system would use a temporary mass storage device to buffer the commercial play, time shifting the output of a cart machine to simultaneously support many program channels.

The spot buffer system is made economically viable by development of low cost mass storage video devices. The characteristics of a good spot buffer are instant random access, or rapid cueing. The buffer storage device could be any of a number of devices such as DRAM, Winchester disk, optical disk, and even, with a slight performance penalty, a videotape format.

The first three have the additional advantage of reducing failures associated with head clogs and electromechanical failures of videotape machines. We are currently pursuing a D2 format disk recording system, which has the advantages of no generational loss in dubbing, and the ability to use digital error monitoring for automatic quality control.

The architecture of the spot buffer system is shown in Figure 1 on the next page. This proposed system has two LMS (main and protect), and uses two outputs from each cart machine. The cart machines would hold 1,000 tapes with a multiple spot format, for an available library of 10,000 spots. Ten D2 disk buffers would play the buffered spots direct to air.

Within the LMS, two VTRs are assigned to play back to each of two output channels. Since spots are not being played to air in real time, there would be no VTR cycle time, or spot conflict resolution problems. VTR cycle time problems occur in direct to air systems when spots are scheduled faster than the cart machine can cue them up. Conflict resolution is required in multiple spot systems where two spots scheduled to run back to back are recorded on the same tape. Both of these timing limitations are overcome by the buffer approach.

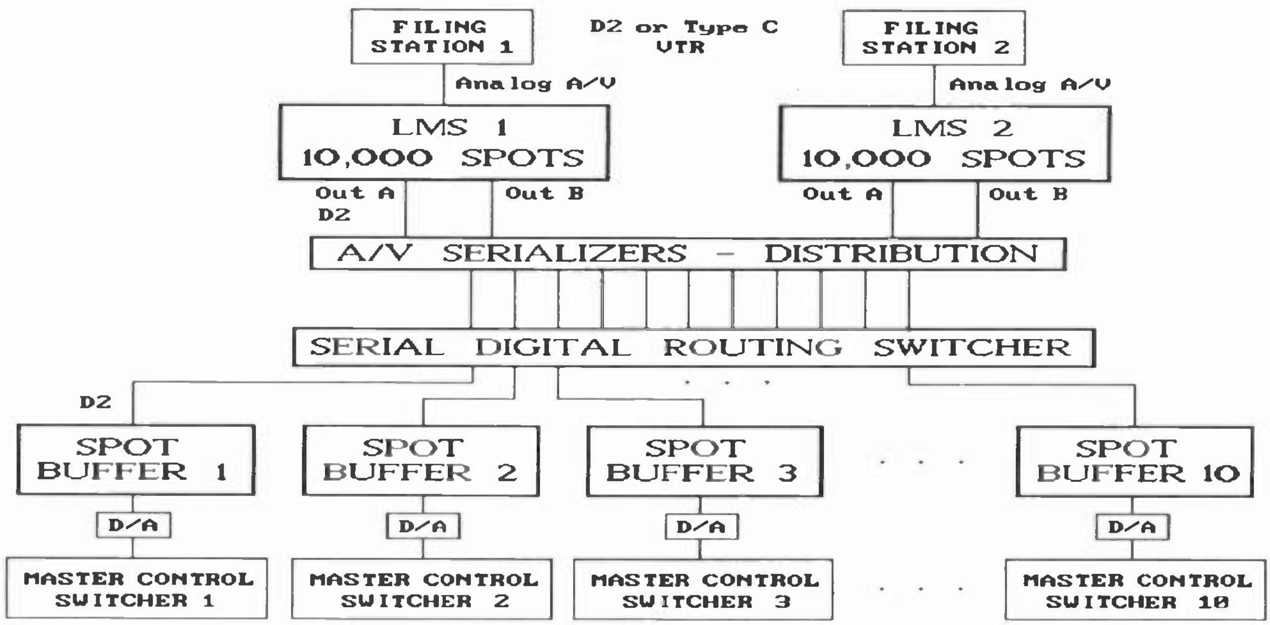

Figure 1. Spot buffer A/V system.

Computer control for the spot buffer system is shown below in Figure 2. This control system would use Local Area Network, RS-422 serial, and parallel control signals to coordinate operation of the spot buffers.

Three Spot Buffer Controller PCs would manage all the dubbing and playback requirements, and a filing PC would manage transfer of new spots into the system.

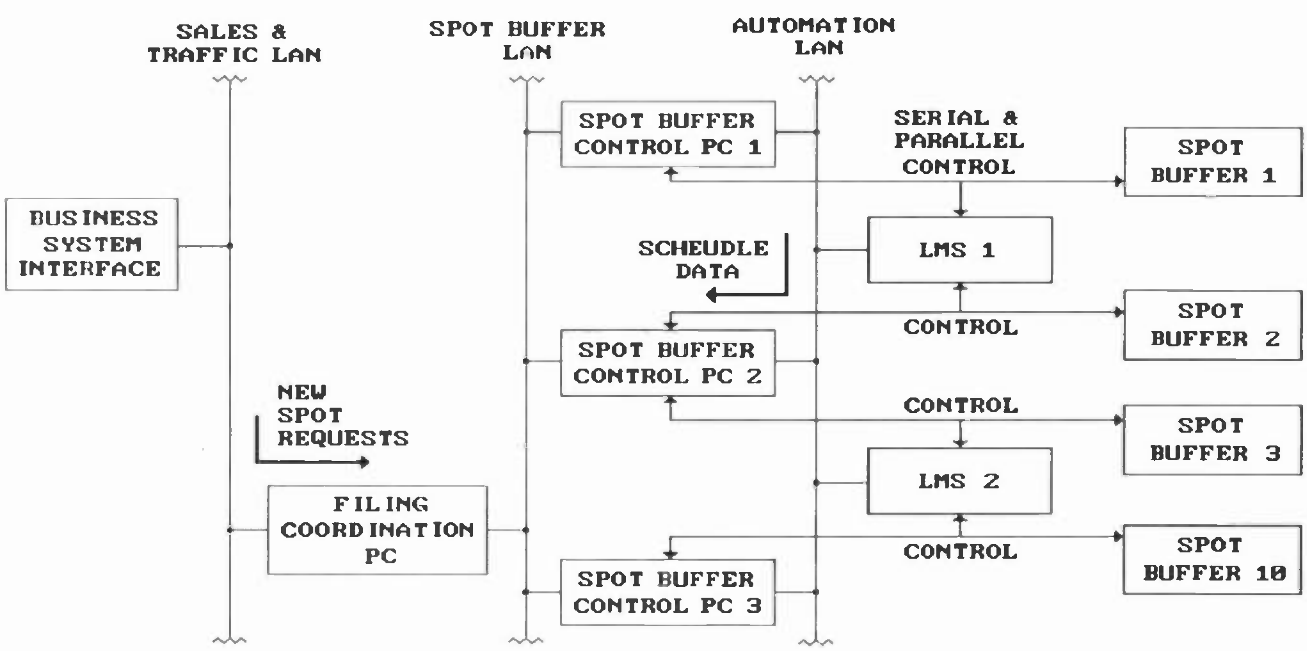

Figure 2. Spot Buffer Control System.

VIABILITY STUDY

The second part of this report presents the results of a study of the spot buffer system. Since the capability, and decision parameters of such a system are not intuitively obvious, CBS commissioned a study by Kerr Vayne Systems Ltd. to find out if such a system would work.

Library Size

The first question examined was how big the spot inventory really was. This was important because historically, the library size was determined by library shelf space available or by tape stock available, and these were larger numbers than would fit into our biggest cart machines.

If the entire active library could be fit into existing cart machines, many operating efficiencies would be obtained.

Methodology

The computer model used to calculate library size tracked spot material from its first use in a broadcast, until it was removed from the system. The parameter hold time gave the time in days that a spot would be held in the library after its last use. The computer model analyzed the schedule for each day and calculated the number of new spots for that day, the number of removed spots, and the total library size. A final category, new blank spots, showed the number of spots which were not assigned an identifying number until just right before use. These spots would typically be daily promotional announcements, and each was counted as a unique spot.

The model was run on the same data for a number of hold periods between ten and 120 days. The model output was formatted and imported into a spread sheet for analysis. A typical spread sheet result for a ten day hold period is shown in Figure 3.

On day one there were 259 unique spots broadcast, out of approximately 700 total. On subsequent days, new spots were added, and the library size grew. There was a strong cyclical pattern of new spots being added on Mondays (days six and thirteen).

Day	New	Removed	Killed	New Blank	Total
1	259	0	0	0	259
2	113	0	0	9	372
3	93	0	0	6	465
4	16	0	0	0	481
5	43	0	0	0	524
6	196	0	0	5	720
7	201	3	3	0	918
8	155	2	2	6	1071
9	139	2	2	13	1208
10	113	3	3	6	1318
11	80	0	0	0	1398
12	67	32	0	0	1433
13	249	58	2	22	1624
14	203	67	1	28	1760
15	182	58	4	20	1925

Figure 3. Spread sheet data.

For a ten day hold time, there were on average 137 spots filed each day. This average dropped to 114 spots for a thirty day hold time.

This is a fairly naive model, since if we know a spot is no longer needed, it could be removed from the library immediately. However, that sales information may not be known in advance, and it imposes additional human management functions on the system, which we want to be fully automated. The library size obtained by this naive model is the worst case, and additional information could decrease the library size.

The main trade-off then was that the library size could be reduced by decreasing the hold time, but you would eventually start removing spots that would be used again, and incur a penalty of repeat filing into the system.

Data

Broadcast schedules for the first half of 1992 (January 1 to July 15) were analyzed. Results are presented here in two categories; total operations (CBS Television Network plus WCBS-TV), which is comparable to a full time network or independent station schedule, and local operations (WCBS-TV), which is comparable to a network affiliate schedule with station breaks and locally originated programming.

Library Size Results

The main results of the library size study are the graphs of total library size verses hold time, and the graphs of filing distribution verses hold time. The graphs below show total library size in Figure 4, and local library size in Figure 5.

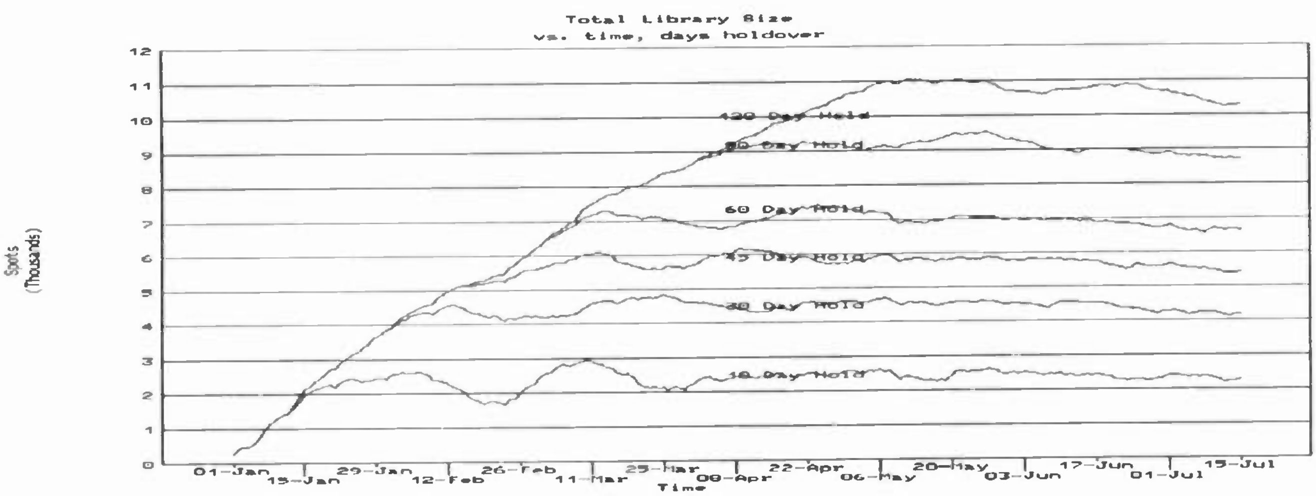

Figure 4. Library size, total operations.

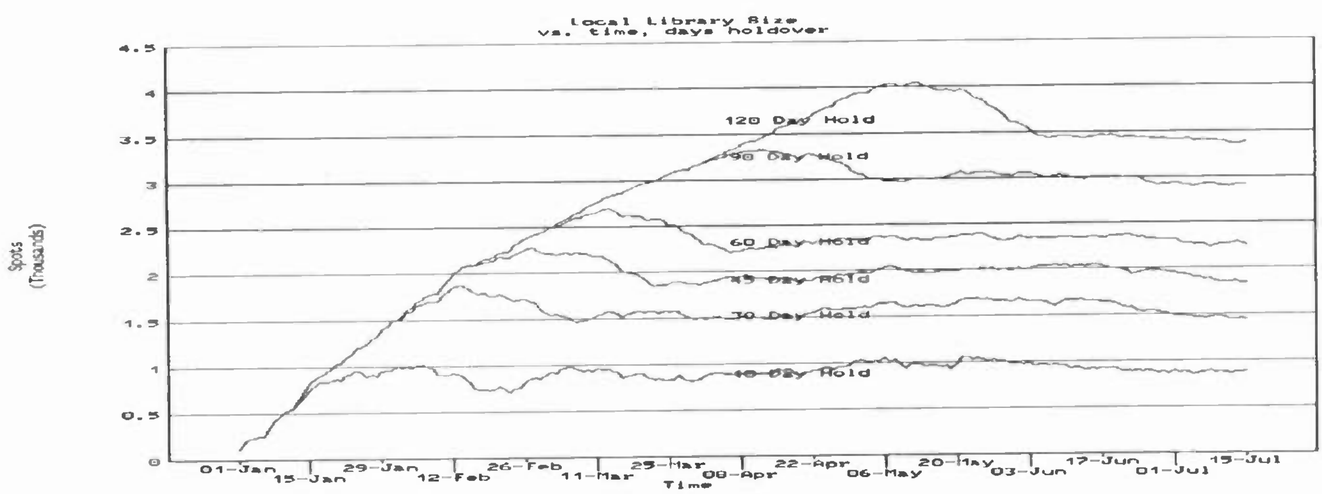

Figure 5. Library size, local operations.

The significance of library size can only be evaluated by next examining the graphs of filing distribution for various hold times. The divergence between these traces shows the effects of premature removal of spots for shorter hold times, which must then be refiled into the system.

Clearly the ten day hold period requires significant refiling, but for the other hold periods, there is no significant difference in the refiling penalty for hold periods between thirty days to 120 days. Looking back at the total library size graph in Figure 4, this means that a library size of 4,500 (thirty day hold) will work as efficiently as a library size of 11,000 (120 day hold). This analysis shows a spot usage pattern that once a spot is not actively used, it tends to never be used again.

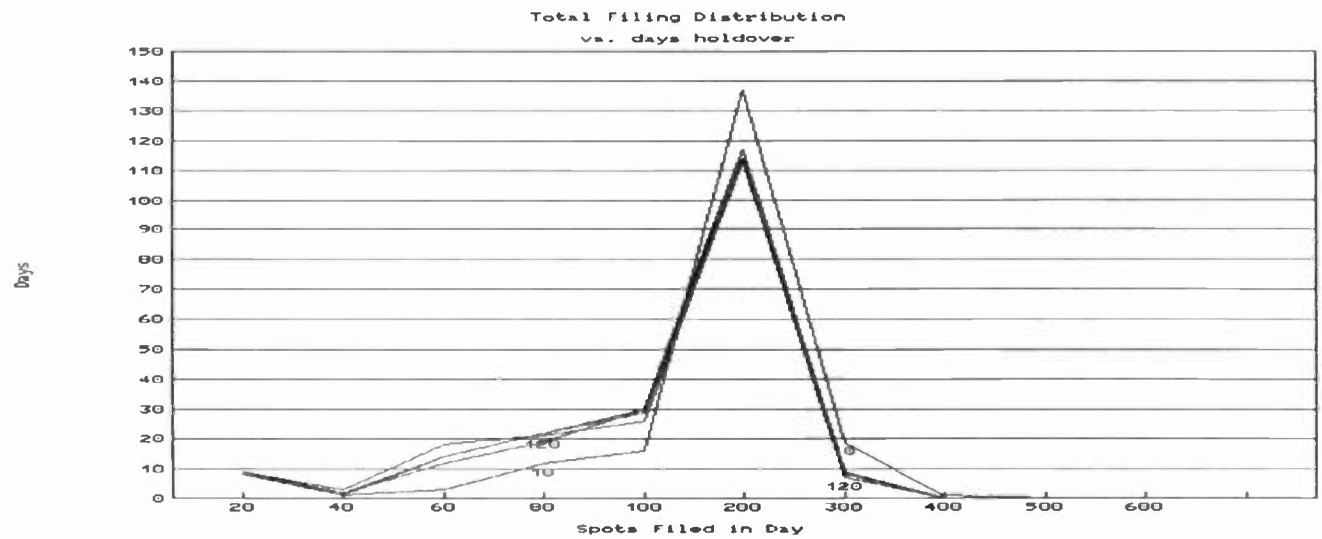

Figure 6. Total filing distribution.

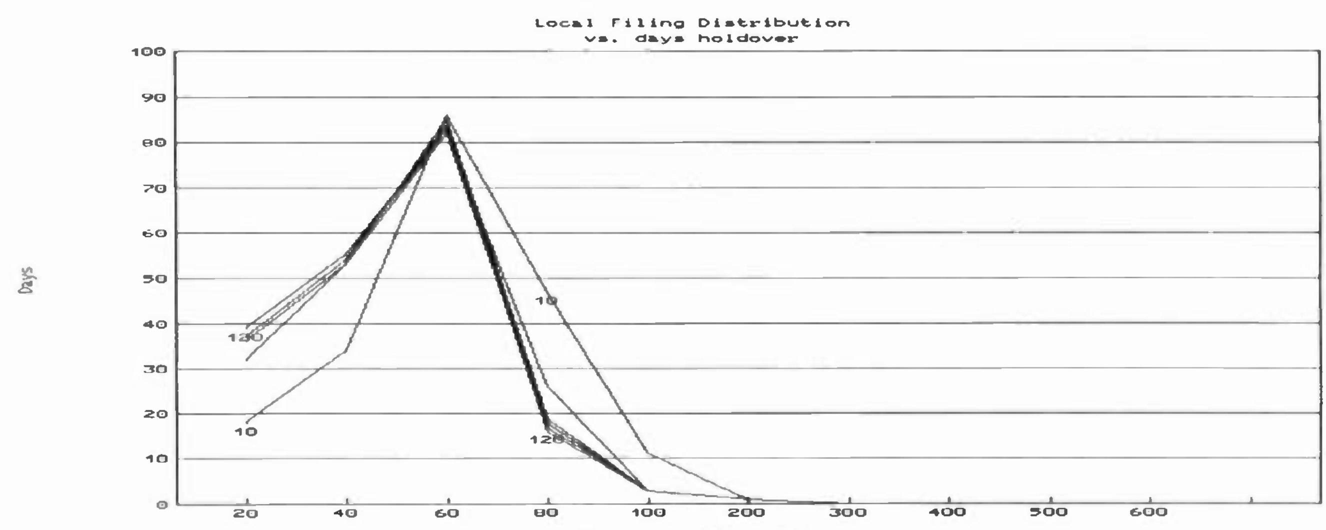

Figure 7. Local filing distribution.

The final results of the traffic study are shown below.

Library Size		
Hold	Local	Network
10 days	< 1,500 spots	< 2,500 spots
30 days	< 2,000 spots	< 4,000 spots
45 days	< 2,500 spots	< 5,000 spots
60 days	< 3,000 spots	< 6,000 spots
90 days	< 3,500 spots	< 7,500 spots
120 days	< 4,500 spots	< 8,500 spots

Average Spots Filed Daily						
	Hold Period in Days					
Operation	10	30	45	60	90	120
Local	50	42	41	40	40	40
Network	96	82	78	77	76	76
Total	137	114	110	108	107	107

Figure 8. Traffic study results.

WAITING LINE SIMULATION

The second part of the study was to investigate various decision parameters of the buffer system design, such as the number of cart machines, the number of VTRs in a cart machine playing back to each output, and the storage time required in the buffers.

Methodology

Queuing theory allows the study of waiting line type problems, where customers arrive to receive service, and queue up in lines to wait for service. These kinds of problems usually are non deterministic. Arrival of customers typically has a random distribution, and service times also have a random distribution.

Queuing theory provides solutions to relatively simple, well defined queuing systems, but for more complex systems, model simulation is used. The problem is described by a software model of sufficient accuracy to provide useful results for simulated operation. Simulation allows us to vary the parameters of the model to study the operation of the system.

For meaningful results, the software model incorporated accurate models of the devices (cart machines, VTRs, LMS elevator, switchers, and buffers) using state machines. Also, the simulation was driven using real schedules and takes derived from on air logs. Finally, a buffering algorithm (a model for control) determined the ordering of dubbing to the spot buffers.

Buffering Algorithm

A workable algorithm had to be devised that would determine the order that spots would be dubbed to spot buffers. Such an algorithm must be simple enough to allow a computer model to be programmed, but powerful enough to be usable in a real system.

Two qualities were assigned to each spot buffer associated with a channel; Minimum Safe Zone, and Channel Priority. The Minimum Safe Zone is the partially filled capacity at which a buffer is considered "safe". The Channel Priority is the ranking of a channel with respect to other channels. A higher priority channel will generally have its spots dubbed before those of a lower priority channel. Both qualities help to determine how the cart machines are used to fill the spot buffers or the order that spots from different channels are dubbed to the spot buffers. Both qualities were parameters for the simulation.

The algorithm determines the spots to be dubbed first by satisfying the following criteria in the given order:

1. In channel priority order, those spot buffers below their Minimum Safe Zone are dubbed until their capacity reaches the Minimum Safe Zone. Within a channel the main buffer is dubbed before the protect buffer on a spot by spot basis.

2. In channel priority order, a spot is dubbed to the spot buffer that is closest to its Minimum Safe Zone. After the spot is dubbed, the partially filled capacity of the buffer will be further away from its Minimum Safe Zone, requiring a reevaluation of which spot buffer will be dubbed next.

As a simple example of how these criteria are used, consider the following three channels each with their own partially filled spot buffers:

Channel	Current Spot Buffer Amount	Priority	Minimum Safe Zone
1	0:30	1 (high)	2:00
2	2:30	2	2:00
3	1:30	3 (low)	2:00

Also consider that the spots to be dubbed in future are all thirty seconds in length. The ordering of dubbing determined by the algorithm would be

Spot	Channel	New Buffer Amount
1	1	1:00
2	1	1:30
3	1	2:00 (Reaches Minimum Safe Zone)
4	3	2:00 (Reaches Minimum Safe Zone)
5	1	2:30
6	3	2:30
7	1	3:00
8	2	3:00
9	3	3:00

During the simulation, the playing of spots from the spot buffers would cause buffer dubbed spot amounts to decrease which in turn may alter the order in which spots are dubbed.

The model also considered that buffer devices could not record and play simultaneously. This is usually the case for disk or tape based buffering schemes. The behavior of the simulation would be to abort a dub to a spot buffer that suddenly started to play, causing a recueing to be necessary in the cart machine before the dub can be restarted.

Data

Data for the simulation model was taken from schedules and as-run logs. The logs were used to determine when spots were actually aired, giving times for manual as well as scheduled events. The schedules were used to determine the order of spots to be played back for each channel. Six channel schedules were run simultaneously using six main spot buffers, with four of these channels using protect buffers. Thus ten buffers were used. In the simulation discussed here, only one cart machine was used with two VTRs to supply results for a worst case scenario.

Buffer capacities were set at fifteen minutes, which is representative for commercially available spot buffers. Buffer Minimum Safe Zones were set at 7:30 for the highest priority channel (main network), 5:30 for the next highest priority channel (regional network feed with greater traffic), and 2:30 for all other regional feeds. The Channel Priorities for these feeds were allocated by the amount of commercial traffic.

The output of the simulator was lines of text for times at which the state of any device (cart machine, VTR, spot buffer, or channel) changed. From this UNIX text tools were used to create files for each spot buffer that indicated the dub margin for each spot, or the length of time between the completed dubbing of the spot to a buffer, and the playing of this spot from the buffer. These text files were imported to a spread sheet for analysis.

The graphs of dub margin over a one week period for a main network channel, and a regional network channel are shown below. The main network channel in Figure 10 has a minimum dub margin of 16.5 minutes during early morning programs, and the less heavily utilized regional channel shown in Figure 11 has a minimum dub margin of thirty seven minutes.

The minimum dubbing margin found was for the main network feed. It remained above fifteen minutes for the complete simulation period. Dubbing margins for other spot buffers with less spot traffic were much greater. Another simulation with two cart machines with four VTRs each improved on these dub margins.

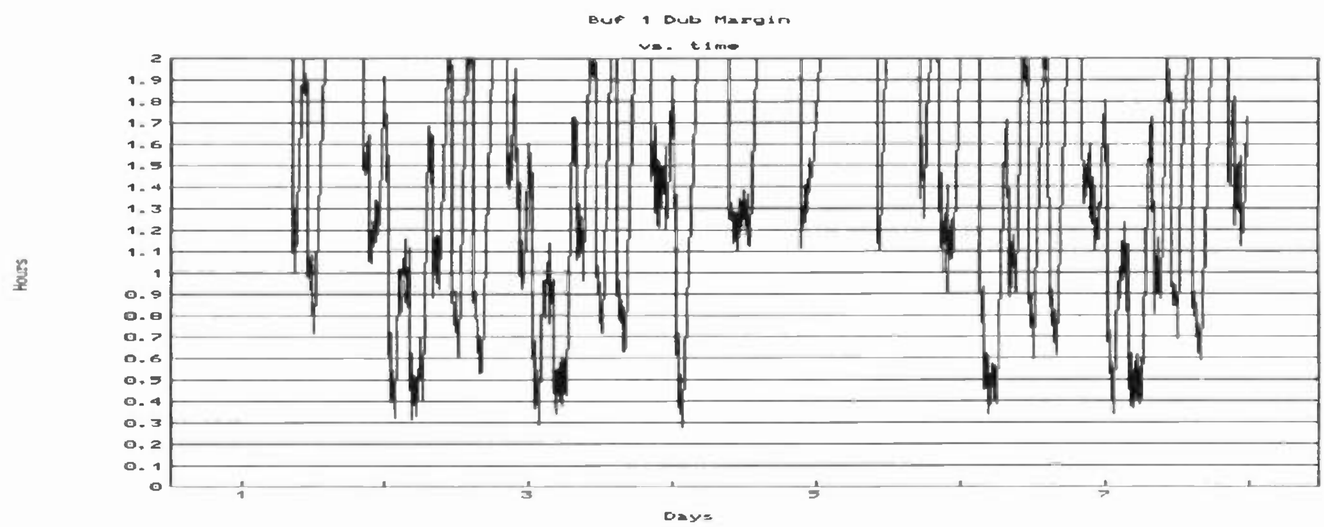

Figure 10. Dub margin, main network.

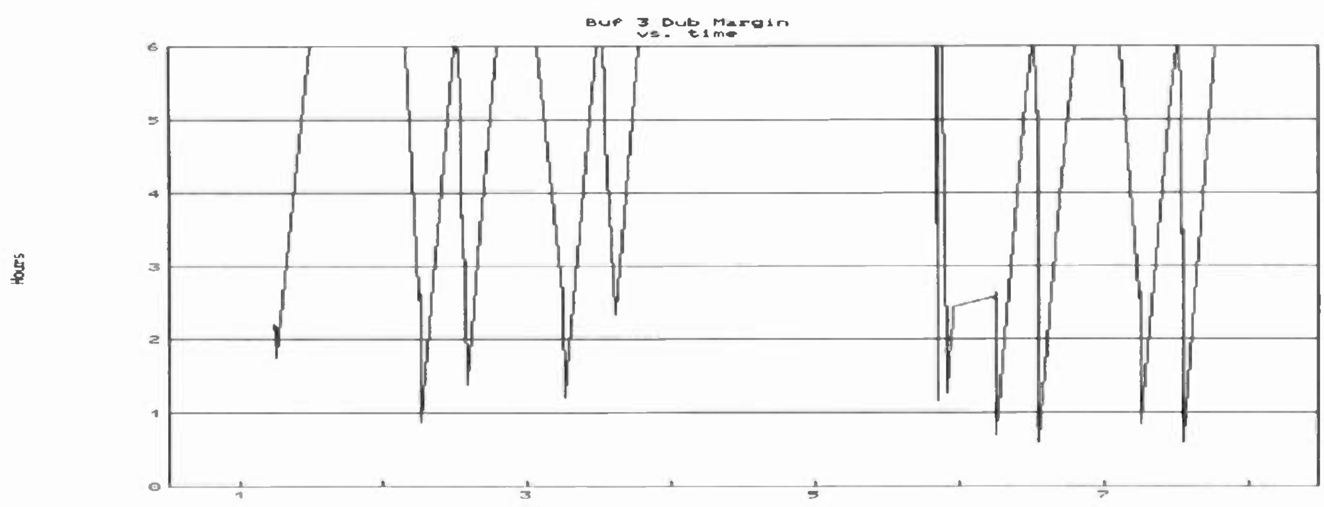

Figure 11. Dub margin, regional network.

CONCLUSIONS

The traffic study portion of the viability study showed that the spot library size does not exceed 5,000 for either network or local operations, given a hold period of thirty days. This spot library size can be accommodated in cart machines currently on the market which offer multiple spots per tape. Having the spot library completely contained in a cart machine gives a reduction in dubbing, movement, and storage of tapes, and the inventory may be managed using tape management functions of the cart machine.

The waiting line simulation portion of the viability study showed that for main and regional network operations, one cart machine with only two VTRs could fill ten spot buffers used for one main and five regional network channels. Using buffers with a fifteen minute capacity, the minimum dub margin was found to be sixteen minutes.

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TV AUTOMATION AND FACILITIES PLANNING

Sunday, April 18, 1993

Moderator:
Charles Dages, CBS Television Network, New York, New York

AN ARCHITECT'S VIEW OF MANAGING TV PRODUCTION FACILITY CONSTRUCTION
Antonio Argibay, A.I.A. and Bice C. Wilson, A.I.A
Meridian Design Associates, Architects
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INTEGRATING A MODERN STILL STORE INTO A PRODUCTION ENVIRONMENT
Bob Pank
Quantel Ltd
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PLANNING WIRELESS MICROPHONE SYSTEMS FOR TELEVISION
Edwin Somers
Audio Services Corporation
North Hollywood, California

AN OVERVIEW OF ONE STATION'S EXPERIENCES WITH ROBOTIC CAMERAS
James Withers
KDFW-TV
Dallas, Texas

DESIGNING CBS-TELEVISION'S FIRST SERIAL DIGITAL, MULTIPLEXEO, VIDEOTAPE EDITING ROOM
Craig Harrison
CBS Television City
Los Angeles, California

AN ARCHITECT'S VIEW OF MANAGING TV PRODUCTION FACILITY CONSTRUCTION

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ABSTRACT

The process used for the Design and Construction of Broadcast facilities has the potential to bring major benefits to the facility owner and the company beyond the apparent goal of building the project at hand.

This paper discusses the role of the architect in the facility design and construction process. It then emphasizes the importance of working within the client organization and between that organization and its consultants and contractors to develop clear common goals and objectives. We then introduce the concept of "Partnering". Partnering is a new term for an old idea; the idea that working together in a non-adversarial team towards the achievement of commonly held goals will result in more innovative facilities with improved quality, reduced costs and more effective utilization of resources.

The paper is targeted both at the experienced facilities management staff which wishes to increase its effectiveness at serving its in-house clientele, and the lone chief engineer undertaking a construction project.

INTRODUCTION - OUR NEW OPPORTUNITIES

We are in a new era in Television Facility design. In the early days of the television industry, people were productive in environments whose design was relatively insensitive to human needs. Those days are ending. We have enough years of experience with both the technology and the basic "building type" of the Television production and post production facility that our vision of them is maturing.

We now know that those facilities which can integrate ever changing technology while truly addressing the needs and functions of their human inhabitants will be the most productive, the most creative and the most profitable.

It is time for the architect, the engineer, the builder and the end user to learn to work together as a team to create facilities which are the ultimate synthesis of systems engineering and humanistic design, to create a workplace which is more than the sum of its technology.

By its very nature this complex undertaking requires teamwork and innovation. In this paper we will offer you techniques which will build better communication and team spirit both within the client company, and between the company and its project consultants.

A typical project has an in-house team comprising multiple end users, broadcast engineers, facilities management, set and lighting designers and management. Once this team decides to undertake a construction project it will usually call on outside expertise in the form of real estate consultants, architects, broadcast system engineers, mechanical and structural engineers, construction managers, general contractors and a myriad of sub-contractors and suppliers.

Each of these players brings to the process their own needs, interests and agendas. The legal structure of the construction industry, its standard contracts and pro forma relationships, are essentially structured on the assumption that these interests are in conflict. There is an unfortunate assumption that "someone must lose if I am to win."

All too often the construction process is destructive and divisive, both within the corporate culture and between the company and its consultants and contractors.

As architects, we understand that it is precisely their individual needs that are the gifts each team member brings to the creative process. The challenge we face in the pre-design phase of a project is to help the team devise and then articulate their common goal. In the design and construction phases we go on to help them interpret that goal into its implications for their interests and responsibilities.

In the process we can help you build a greater cohesiveness and understanding within your company and between the company and its consultants and contractors. The Partnering process will help you utilize the planning phase of the project for the refinement and clarification of long term business goals and objectives.

Innovative facilities created with minimized risk, improved quality, reduced cost, and more effective utilization of resources will result from the application of the management and human relations methodologies described in this paper.

A HIGH RISK UNDERTAKING

The Construction of a Television Production or Post-production facility is a very complex technical undertaking, absorbing considerable human resources, capital and material. In undertaking a construction project you are essentially creating, within your company a new temporary business venture, which we refer to as "The Construction Enterprise". This entity will be responsible for planning, designing and constructing your own working environment. This is, by its nature, a high risk endeavor. We will describe some of those risks, and the Partnering strategies which can turn them into opportunities.

Construction is not your business: Any business start-up is risky. The risk to your company when you are starting up a business within a business and utilizing people who are placing their normal careers in jeopardy is even greater.

You owe it to those responsible in-house for the project to take this new business start-up as seriously as you would a new venture. Create clear goals and responsibilities. Consciously build communication systems tailored to the enterprise, and bring in, at the earliest possible date, whatever outside consultants may be required to flesh out your team and assure its success.

New Relationships are created: Failures in the design and construction process often stem from failures in the human relationships among the members of the project team. For this effort, you will bring in outside consultants and contractors. You will also ask your staff to step away from their day to day activities and assist you in devising the Business Plan and Architectural Program which will be the basis of the design. People who previously were colleagues collaborating on their work are now needing to negotiate priorities for the allocation of limited resources of space, capital and professional services.

Take the time to carefully build those new relationships so that this negotiation can be positive, overt, and take place within the context of clearly defined common goals and a viable system of decision making.

Everyone's got Money on the Line: Each party to the construction process has money on the line. Whether it's the department that will devote its hard won capital budget to the process, the architect whose profitability and reputation depends on how well the whole team works together, or the contractors who have bid on the project, all are interdependent for the health of your respective bottom lines.

Conventional construction relationships are oriented to adversarial resolutions of this interdependence. This discourages risk taking, innovation and cooperation and magnifies the negative impact of any dispute or of a mistake on the part of any team member.

Partnering based relationships are dedicated to equity in meeting the recognized needs of each party. In such a setting responsible risk taking, innovation and cooperation are encouraged and problems become a means for bringing the team closer together in their focus on the common objectives.

Problems will arise: Each construction process is an act of faith. Faith that a team will come together, understand your needs, manifest them in a design, and build that design within your budget and schedule. Faith that the disparate team which coalesces around a project will be able to work out their differences when the going gets tough.

> Even with the most seasoned and professional team at hand, problems will arise. It is foolhardy to trust in faith to see that the team will work well together to resolve those problems. If the team is new, apply Partnering techniques to build their problem resolution skills. If you start to build a good team, and go on to do other projects, don't throw away the valuable experience invested in them.
>
> One of the key ideas behind Partnering is the importance of a Long Term Commitment to building a team. At the end of a project take the time to assess your successes and failures. Discover ways your processes can be refined for next time. Partnering has immediate benefits on any project, but its greatest benefit occurs after the team has been through the drill several times.

Each of these risks is an opportunity to enrich the design process.

THE WONDERFULLY EMPIRICAL

There are aspects of the design process which are wonderfully empirical (or at least nearly so). From lighting and glare issues, to console design and carpal tunnel syndrome or HVAC system design and issues of ionization of the air by electro-magnetic fields and their effect on productivity, many of the formgivers for a facility are objective and scientific.

With adequate research and experience in the creation of Television Facilities any competent professional should be able to assist you in establishing a set of performance criteria, and to then embody them in a design. However, just making correct choices in this regard does not assure you of a well designed, or a well built facility.

WHAT DOES AN ARCHITECT BRING INTO THE PROCESS?

An architect brings experience and insight into an undertaking which may be foreign to your organization. The architect can assist you to synthesize and balance all the compelling and often competing form givers which would shape a project, and to find the most elegant possible manifestation of their synthesis. We use the term elegant in both its aesthetic, and its scientific sense. In scientific terms, an equation is elegant if it is the briefest and clearest possible expression of a set of conditions.

Architecture is one percent inspiration and ninety nine percent perspiration. The pleasure of achieving this elegant solution is that one percent which makes all the perspiration worthwhile. It is a somewhat intangible skill, yet it is based on a careful and tangible methodology.

This methodology is fundamentally the ability to help the team to build, both within the client organization, and among that organization and its various consultants and contractors, a sense of common purpose, of clear goals. Our responsibility is to help all the competing interests at play in a project to work as a team. We can then create a means for each member of that team to effectively communicate their needs and to then effectively play their role in the facilities creation process.

One of the key concepts in our design methodology is that, if the concepts are clear, the details will follow. Conversely, if there is no clarity within the project team on the concepts or core goals which underlie a project the details will never fall into place.

Using Partnering you will build a clear consensus within the team as to the goals underlying the project and have an effective process at hand for implementing those goals.

When the Project Team has been successful in this undertaking, the facilities construction process becomes most cost effective, and efficient. It can then play a powerful role in building a healthy and productive corporate culture which lasts long after the paint is dry.

In recent years this approach has become known as "Partnering". The following is an introduction to Partnering, and a description of its potential impact on the Construction Enterprise.

PARTNERING

We have been using Partnering concepts in our practice with great success since 1984. It is an idea as old as the handshake and as new as the realization that the teamwork required to create a good facility is an opportunity, not a liability.

Partnering is defined by the Construction Industry Institute as:

> ". . . agreements between companies to cooperate to an unusually high degree to achieve separate yet complementary objectives . . . . a long term commitment between two or more organizations for the purpose of achieving specific business objectives by maximizing the effectiveness of each participants resources. This requires changing traditional relationships to [develop] a shared culture without regard to organizational boundaries. The relationship is based on trust, dedication to common goals, and an understanding of each other's individual expectations and values. Expected benefits include improved efficiency and cost effectiveness, increased opportunity for innovation, and the continuous improvement of quality products and services."[1]

In a healthy construction process the natural evolution of relationships is towards Partnering. The design and construction processes are inherently a team process. Broadcast Facility design is even more so.

The 8 Key Elements of a Partnering Relationship:

Partnering can be applied to projects of any scale or duration, with the level of formality and effort required increasing with project scale. No matter what the scale, there are eight key elements of an effective Partnering strategy.

I. Long Term Commitment -

In order to feel safe taking the risks involved in the facility design and construction process team members must feel that there is a commitment among the team to build relationships and resolve disputes equitably. The team must have confidence in this commitment in order to learn lessons from their experiences.

Partnering is most effective if this commitment transcends the duration of any single project, and includes the agreement to assess the effectiveness of the team's performance at the end of the project, and then to apply the lessons learned on the next project.

II. Trust -

"Teamwork is not possible where there is cynicism about others' motives. Through the development of personal relationships and communications about each stakeholder's risks and goals, there is better understanding. With understanding comes trust and with trust comes the possibility for a synergistic relationship."[2]

An atmosphere of Trust must pervade the Construction Enterprise. Team members can then combine resources and knowledge without concern for adversarial conflicts.

III. Shared Vision -

Positive compromise and synergy can only take place in the context of a vision or goal shared by the entire team. This shared vision offers a non-adversarial context in which to come together.

The effort required to understand and synthesize each others goals and expectations will be re-paid manyfold throughout the project.

IV. Equity -

Trust is only possible in a relationship founded in equity and parity. This requires a basic commitment to understanding each others expectations, and to establishing parameters on those expectations which all agree on as equitable.

The traditional "Win-Lose" adversarial relationship must be replaced by a commitment to "Win-Win" solutions.

V. Systematic Implementation with Continuous Evaluation -

Partnering involves fundamental cultural change for most organizations. Tradition and organizational inertia are major impediments to its success.

The team must work together to agree on a strategy for systematic implementation. This strategy must include a mechanism for continuous evaluation and refinement of the effectiveness of the strategy. It is likely that a company new to the concept may need to jointly attend a Partnering Workshop during the project start up phase.

One key element of this strategy must be a process for the timely responsiveness to questions and problems. Such a process can keep problems from becoming disputes. Issues should be resolved at the lowest appropriate level of the team. If the resolution runs into a blockage at any level there should be a mechanism for its timely escalation to the next level of management.

VI. Mutual Rewards & Shared Risk

Partnering works best where performance is encouraged by mutual rewards in light of each party's investment and the risk undertaken.

At the same time, risk must be shared amongst the team based on the relative investment of each member in the project and the reward each member expects upon successful completion of the project.

VII. Synergy

An environment must be created which encourages integrative thinking with the open exchange and consideration of ideas. With the combining of resources and knowledge base, innovation is supported and the whole can become greater than the sum of its parts.

VIII. Build Competitive Advantage

In the team context each member is free to fulfill the role they serve best, supported by their team mates. Each has an equity in building the competitive advantage of the team as a whole. By working together to lower costs and increase the quality of their product they assure the long term commitment on the part of their clients and each other to the future of the team.

Implementation of a Partnering Relationship:

In this section we will look at how Partnering can be implemented in a typical "Design-Award-Build" project delivery method. Other methods of project delivery, such as, "Fast-Track" and "Design-Build" also benefit from Partnering with some modification to the implementation recommendations below.

There is a natural tendency for a successful Partnering Team to evolve towards a Design-Build-Assess process in which the builder is involved with the project from the outset. The process becomes cyclical rather than linear.

This section will give the reader a sense of applicability of Partnering in each of the typical phases of a project.

Pre-Design Phase

The first place to apply a Partnering methodology is during the process of devising your Business Plan for the new facility. It is not fair to assume that the complex team within the client company that will be using the facility, overseeing its design and construction and then maintaining it, can effortlessly come together on an Ad Hoc basis, meld their individual interests and fall into place behind a common objective. There must be a pro-active effort on the part of those responsible for project management to:

- Build a sense of coherence and commitment among the members of the in-house project team.

- Build an atmosphere of trust to eliminate adversarial relationships. Develop open recognition of the equity each player has in the project.

- Make a commitment to negotiate conflicts with an eye toward devising a "win-win" resolution.

- Empower each team member to feel free to invest in the project those strengths they uniquely possess.

- Develop a consensus as to the mutual goals and objectives the team is committed to achieving.

- Agree on a process for the implementation of these goals. This is your Business Plan for the Construction Enterprise. Once the Business Plan is well underway, bring in your architect for his assistance in translating your Business Plan into an Architectural Program. There will be an ongoing feedback loop between these documents and the design process until the elegant solution mentioned above is found.

- Commit to the ongoing evaluation of the success of your implementation. Encourage constructive criticism of previous projects and facilities, both as to their processes and their products. Develop an institutional memory of effective techniques as well as pitfalls.

Schematic and Design Development Phases

These phases define the physical solution to the goals outlined in the Business Plan and architectural program developed in the Pre-Design Phase. The process is one of synthesis of the program with the site and other local conditions of a project.

Design is also the phase where compromises are made among the competing needs of the team members. If the goals and protocols of the project are clear, this process can be very positive as it clarifies the true needs and priorities of those involved in the project team.

The important aspects of implementing Partnering in the design phase are:

- Assembly of the remainder of Project Design Team. Integrate them with the Partnering process already underway among the in-house team members. Maintain active involvement of "Users", "Engineering" and "Facilities" in the design process.

- Building project protocols regarding information formats, and flow of project data and decisions. CAD technology, Computer Networks and E-Mail can be invaluable in this regard. Minimize paperwork, maximize exchange of important data. Encourage team members to optimize the utility of the information created for the project. For example, delegate responsibility for documenting existing conditions to a specific sub-set of the team. Another example is planning ahead for the eventual use of your project database for Facilities Management purposes.

- Encourage open communication between different members of the team to encourage innovation and resolve problems synergistically. Initiate regular informal reviews and brainstorming across disciplinary boundaries. Don't wait until the "Big Meeting" to get feedback and develop support for an initiative.

- Balance the responsibility of the architect to serve as the "orchestra leader" or facilitator within the team with recognition of the need for mixed-team leadership aligned with responsibilities and expertise.

- Pre-qualify prospective builders and suppliers of the various trades and introduce them to the project and to the methodology of Partnering at the earliest appropriate moment. Involve them to determine costs, availability and compatibility of their products with project goals.

- Follow an orderly process of moving from design concepts to ever smaller levels of detail.

- Create a "no-fault" context which encourages problems and conflicts to surface and be addressed as early as possible.

Bidding & Negotiation Phase

Assuming that your contractors are not yet on board, this phase will see the third leap in the size of the project team.

To optimize the benefit of a Partnering approach we recommend the use of Lump Sum bids leading to contracts based on a Cost of the Work Plus a Fee relationship. This contract form keeps the contractor's books open to the other members of the project team, and allows greater confidence and openness of communication.

Activity in this phase includes:

- Inform bidders that they will be expected to participate and invest in the Partnering Process for the project. Determine the extent of their interest in working in a Partnering relationship.

Construction Phase

Nowhere in the contract does it tell the team how to build the project successfully. It just specifies the intended result and establishes minimum performance criteria. Now is your opportunity to develop, with the input of all responsible parties, a specific Partnering Implementation Strategy to make the project a success.

- Once a preferred bidder is selected and a contract signed, set up a series of meetings specifically dedicated to building Partnering skills and laying the basis for working relationships. As many as possible of the complete project team, including in-house personnel, consultants, the contractor, his staff and sub-contractors should attend.

In a larger project this may be a retreat for several days with a professional facilitator, and may result in a written, non-binding Partnering Charter. On smaller projects this may be a meeting which results in a Project Memorandum describing the protocols decided. What is essential is to get commitment from all the players, and to build the atmosphere of Trust, Equity and Common Goals.

Devote time to building relationships outside the crucible of crisis. Don't wait until you sit down together to negotiate a solution to your first problem.

- Now that the basic project goal has been expressed in a design, explain that goal to the contractor, and then reframe your goals in terms of the construction process. Be open to feedback from the new team members about alternative construction means and methods which will fulfill your basic goal and increase the competitive advantage of the team.

- Update your project protocols for information flow and decision making processes. Develop a joint project schedule. In setting up protocols, look for opportunities to minimize paperwork, while using information flow to serve the real needs of the project. Emphasize the expedient flow of submittals and shop drawings.

- Set-up explicit conflict resolution protocols. Encourage solution of problems at the lowest appropriate organizational level. Emphasize the importance of timely resolution to issues, including the escalation of the resolution process up the organizational ladder if it gets stuck.

- Utilize Alternate Dispute Resolution strategies for intractable problems. The initial arbitrator of disputes in the construction process is the architect.

- Make time to regularly step away from the work and refine effectiveness of Partnering Implementation Strategy.

Post Construction Phase

It is the level of attention paid to this phase which will make the difference between building a stable effective team with a useful institutional memory, and having wasted all the effort and lessons to be gleaned from this completed project.

- Take the time to debrief. Analyze the strengths and weaknesses of all parts of the project, from materials choices and design decisions to human dynamics. If you have been successful in building the Partnering Process with the assistance of your Architect you should still have the atmosphere of no-fault openness that allows for candid discussion.

- Few people realize that most Construction Contracts establish a one year warranty on all parts and labor of the finished product. Even if you don't go on to do other projects together, maintain your relationships with the General Contractor and his sub-contractors. Use their input to fine tune the project and its sub-systems.

- Work with your team to assemble a package of project documentation and owner's manuals for the systems and equipment. Your architect can assist you in setting up a facilities management database if the project was documented in appropriate media. Carefully archive this documentation. It's incredibly valuable.

- After the facility is commissioned and has had its shake down period, meet with the facility users in-house team members and your design team to get user feedback which can help shape the next project. Document this feedback so that it can survive in the institutional memory.

<u>Potential Problems Implementing Partnering -</u>

Partnering is a management process. It requires fundamental cultural change on the part of individuals and companies accustomed to thinking of the construction process as an exploitive, warlike undertaking.

Success will only come if there is strong commitment at all levels of the company to support this change and implement the process. This requires a commitment of time and resources. It also requires an atmosphere which recognizes that such change will take time, patience and perseverance to bring to fruition.

The issues which arise during a typical Partnering start up project are all related to the newness of the idea and the need to tailor its principals to the specific people and companies on the team. Typical issues include:

- Lack of time devoted to pro-actively building relationships.

- Initial awkwardness in articulating team objectives and underlying concepts.

- Rigidity and resistance to change on the part of team members.

- Treating Partnering as yet another source of paperwork and procedures rather than as a process aimed at streamlining the accomplishment of project goals.

- Cynicism within the organization which treats the process as a passing fad rather than a fundamental change.

- Partnering is a process of investing in people and organizations the effort required to shape them into an effective team. This shaping and its related changes can be scary. In the new spirit of openness hidden weaknesses in organizations can come to light which may be awkward in the short term while pointing the way to long term strength and competitive advantage.

IN CLOSING

Our goal has been to offer Television Facility owners tools which will optimize their next construction experience. If the ideas touched on make sense to you, our bibliography offers many resources for further study.

Too many organizations approach the construction process with a sense of dread and foreboding. This is a shame. With the help of competent professionals and the implementation of the Partnering process the Construction Enterprise can be an enjoyable, creative undertaking. It can strengthen your company and help to clarify its fundamental goals and long term planning. Good luck on your next project.

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INTEGRATING A MODERN STILL STORE INTO A PRODUCTION ENVIRONMENT

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ABSTRACT

Digital technology has been able to provide much more for still stores than just a replacement method for airing whole pictures. The use of linear keys, to present cutouts as floating graphics over backgrounds, has led to some complexities in operation and artifacts. More complex interfaces, special keyer processing and heavy use of the switcher have been proposed as remedies. A different approach, placing more processing in the output section of the still store, re-establishes an easy interface and simple method of operation.

INTRODUCTION

Still stores have come a long way since their arrival in the early 1980s. The first motive was to replace slide scanners and, as such they could be used in the production studio in a similar way. Now there is a range of models available on the market, some still aim to bring that basic functionality at a good price, others, making more use of the opportunities that digital technology can bring, offer much more. To get the best from still stores they need to be used and applied in a fitting way. That includes interfacing them appropriately into the studio.

Recent trends have been to use a dual output configuration to provide preview and main feeds, and for the display of cutouts as floating graphics. In some cases the output pictures may themselves be montages assembled from a selection of cutouts. As much as the style is liked there have been technical difficulties in achieving a clean result when the cutouts are keyed over a background. Some solutions offered give improvement in certain areas but involve a heavy use of production equipment which, apart from considerations of cost, puts more strain on the already pressured environment of the on-air switcher and its operator. The total cost may wind up being more than that measured in dollars as the opportunity for errors multiplies! The efficient, straightforward method for airing stills, that had been established in earlier days, is lost.

The technical difficulties encountered may appear to be a consequence of a combination of factors, some from the still store itself and some from the use of its signals downstream in the keyer. If established techniques are applied to the use of the cutouts, for the floating graphics style of presentation, the keyed result appears with a halo surrounding the cutout and there are problems with making cross-fades . The still store/switcher operation which uses a combination of normal cutouts applied to a standard linear keyer does not give the required results. There appears to be something wrong with the interface. This paper reviews the interface concepts already used and shows how the aims of regaining the simplicity and efficiency of the old style of operation, as well as taking advantage of the new style presentation, can both be achieved.

THE ESTABLISHED STYLE OF OPERATION

By the mid eighties there had developed a way of working with still stores that gave good results and was easy to use. Still stores are established as the method for storing and presenting pictures for use in TV productions. At first their applications centred on live programming but now they are used widely in many areas of production, including post.

The simplest use of still stores is to record and present whole pictures. In this operation they may be used as a video device with a straight input and output. Such a system would comprise a disk for bulk storage of pictures and a framestore for holding the video input and output picture (see figure 1).

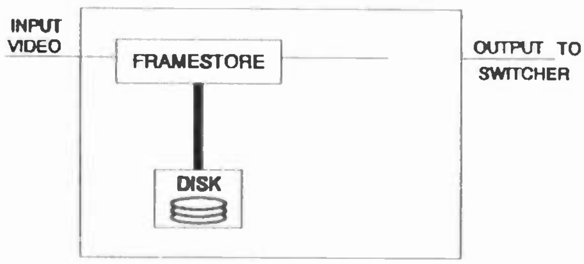

Fig. 1: Single output still store

The method of transition from one still to the next is governed by the data transfer from the disk to the framestore. This would most likely appear as a wipe down the screen at a speed dependent mainly on the data transfer rate of the disk typically in the order of a second. This may suit the requirements for standby captions or for stills in an edit suite, but is of little use for most live operations.

The next stage is to introduce the dual output framestore. (see figure 2).

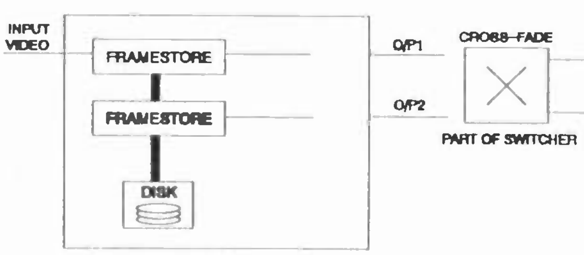

Fig. 2: Dual output still store

In this both the on-air and the next pictures are loaded and available as two separate outputs. An external switcher can be used to make the transition, typically a wipe, cut or dissolve, from one still to the next. At the completion of the transition the next still is loaded into the off-air framestore, and so on.. This type of operation is easy enough to understand but is not ideal. For one thing the on-air and next video sources are swapping after each transition requiring the stills operator to think more carefully each time a still is changed. At the same time the switcher operator also must check which way around his still sources are before making the next transition. Such a situation increases both the load on the operators and the possibility of error.

In many cases operation is eased by the use of both more automation as well as additional facilities in the still store. Stills can be stacked to appear in the required order and loaded into the correct framestore. By the addition of a combiner (see figure 3) the picture transition, cross-fade or cut, can be made within the still store.

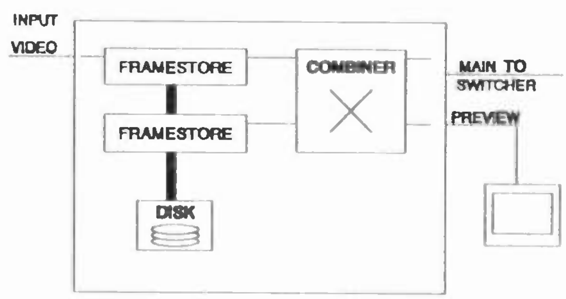

Fig. 3: Dual output with combiner

This can be arranged so that only one output is ever used as the main. The prescribed sequence of stills will appear complete with its chosen transition at this output with the second always acting as a preview. This arrangement is the easiest to accommodate and safest to use since there is always only one source to be accepted from the still store, with all the sequencing and transitions taken care of, and a clear preview of the next picture to come in the sequence.

OPERATING WITH HARD KEYS

Key outputs were introduced with the early still stores for use with their special effects. This amounted to the cropping, re-sizing and bordering of pictures. In general the images remained rectangular, except in the case of montages but there the critical factor of the edges remaining only vertical and horizontal was maintained, and so a hard key was sufficient to give good results when keying the output over a live background (figure 4).

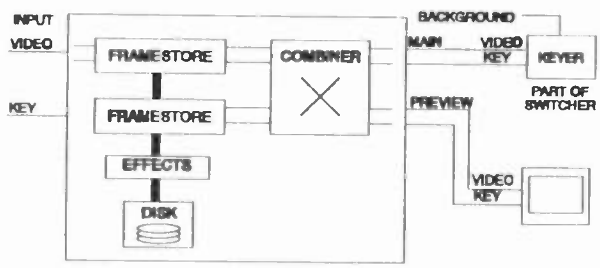

Fig. 4: Dual output with effects and key

In this way the now well established over the shoulder still for news presentation can be easily keyed and all the transitions taken care of by the still store without any extra operation from the production switcher. This applies provided that a set of rules is followed.

Cut transitions to any size or position can be made with no problem but for dissolves successive stills need to be the same size and position otherwise the necessary sudden change in the hard key would spoil the effect. Likewise changes between reduced size and full sizes would not look correct at the keyer output. The way around this is to use more of the switcher for source selection, two keyers and a cross-fade of their result. In some cases this will occupy all three mix effects banks of the switcher which seems overspending for what should be the simple task of implementing a stills presentation.

USING LINEAR KEYS

Modern stills presentations require cutouts as well as whole pictures to be output. With the cutouts the key signal needs to be linear. When applied to a linear keyer this allows the irregular cutout shapes to be keyed with smooth, or even soft, anti-aliased edges as well as enabling the presentation of transparent foregrounds. There are extra demands, pitfalls and new situations to be taken care of. These can be handled but it is necessary to adopt the right techniques.

The storage and use of linear keys makes extra demands for the use of stills.

1) The stored image must include its associated linear key signal. For those using the CCIR 601 4:2:2 component digital coding standard this will mean the use of a 4:2:2:4 system.

2) The provision of key input and output for the still store must be a part of the installation.

The use of linear keyers does require some extra care, for example they must be correctly set up for range and gain otherwise there may be artifacts such as the foreground appearing to be transparent where it should be opaque. Digital keyers are ideal for this as they do not drift and will operate exactly. This becomes even more important when operating in a component or RGB environment where all channels should track each other precisely.

CROSS-FADES

A linear keyer operates so that the level of its video output, foreground A and background B is controlled linearly by the level of the key signal K so that the output is

$$A.K + B(1 - K)$$

Since K can be any value between 0 and 1 it means that the transition between foreground and background can be a precisely controlled dissolve. Hence its ability to present anti aliased edges and transparency. The linear keyer can be regarded as a pixel rate fader and, as such, can be used to dissolve foregrounds over backgrounds. If this is the case then a large part of switcher operation that would otherwise be devoted to this function can simply be assigned to the keyer. If it were possible to make a cutout to cutout transition this way there would be an important benefit in that switcher operation could be greatly simplified.

The level of the key can be used to control the opacity of a cutout over a background. One way that a cutout to cutout transition could be displayed at the output of a linear keyer is shown in figure 5.

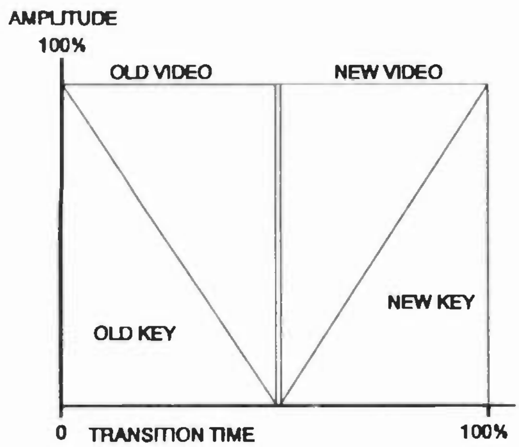

Fig. 5: Fade out/fade up transition

The outgoing cutout is held at the output for half the transition while the key fades to zero, then the cutout is switched to the new one and the key faded back up. The keyed result shows the old cutout fading out and the new one fading up giving a useful type of transition with only the use of a linear keyer.

It is not so obvious how to achieve a cross-fade in such a set up. Figure 6 shows what happens to the outgoing cutout X and the incoming cutout Y. It would seem it should be possible to achieve a cross-fade transition by fading down the video and key for X while fading up those for Y. It is clear that the start and end points of the transition will be correct, but in between things are not right. Assume that both X and Y are white (1) cutouts presented, as usual, on a black background. The background video, B, to be added in the downstream linear keyer, is also white (1). If the system was perfect, the output of the keyer would stay totally white throughout the transition. There are three types of area to consider: those that always remain background, those where the cutouts overlap and those where only X or Y ever exist. The first areas present no problems as there is no change during the transition. Consider the mid point of the transition. In the overlap areas the key level would be 0.5 for X and 0.5 for Y making a total of 1. There the video would also be a 50-50 mix of X and Y and the total keyed result would be correct. In the non-overlapping areas where only X or Y should ever be displayed, the key level will be at 0.5 and the video will be a 50-50 mix of X or Y and black; i.e. one of the cutouts and the black surround from the other. As a result, after being multiplied by the key, video in this area is seen as darkened by 25% at the output of the keyer.

At this point a fourth type of area can be mentioned, transparent areas. Following the same logic as above these will also not look correct during a transition.

Clearly this method does not work but cross-fades remain an important facility for production. It would be very useful to maintain the straightforward operation with the single linear keyer but some other form of signal processing would be required. There is the obvious solution of using two keyers and a down stream cross-fader but that solution is not elegant, uses too much equipment and is over complicated in operation. Besides this the linear keyer itself is not able to produce the right results for keying a single cutout (see below)!

HALOES

The cutouts are a creation of the key signal which has already been applied to the original picture. If this cutout is placed at the output of the still store then the same key signal will be used again for the keyer. This is effectively using the same linear key twice and the upshot is that it does not give the right result. Internally the still store places the cutouts over a black background and, at the output of the keyer, there is a halo of that background showing around the cutout, when it is placed over the background video. This is the same process that causes the darker areas in cross-fades. In some cases this is seen as a useful effect as it helps to highlight the edges of the still but more usually it is considered an unwanted artifact. There have been a number of early approaches to the elimination of this problem but they each have some significant drawbacks.

With the introduction of linear keys the problems of the halo effect and the difficulties with cross-fades spring up. The ideal solution will not only solve these but will allow simple operation and not demand the use of large areas of production equipment.

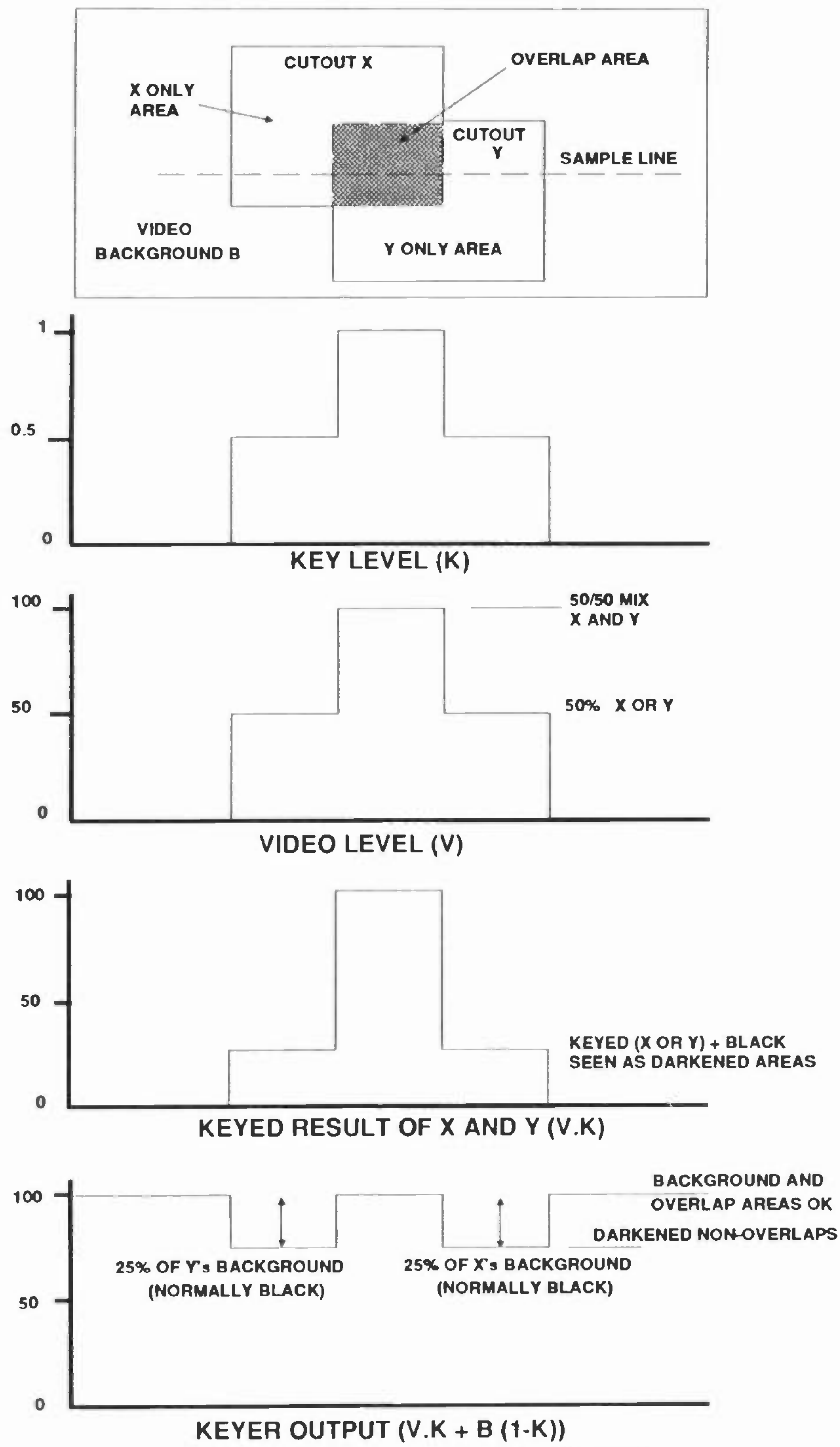

Fig. 6: Mid cross-fade of keyed cutout

Solution 1: Use whole pictures

The first and perhaps most obvious alternative would be to output the whole of the original picture, or cut rectangle within which the

cutout exists, so that the key signal is only applied once, in the linear keyer itself. Technically this gives the correct result but it does introduce some serious operational complications.

1) The only way to see a preview of the cutout would be to look at the output of a linear keyer.

2) Since the plan is to use the key signal only once, at the final linear keyer, there would be only very limited possibilities for using montages assembled within the still store or Paintbox. This would be a major restriction.

3) To create dissolves two linear keyers are needed with a cross-fader at their output. This goes back to the overspending situation, already outlined, where a large part of the switcher is tied up with the presentation of stills.

4) Attention would need to be given by both the stills and switcher operator because of the constant swapping of preview and on-air outputs.

Solution 2: Use compensated keyers

It could be viewed that the problem appears to be in the keyer itself. After all, the signals going in are correct, it is the output that is not! Compensated linear keyers have been produced and they can solve the problem of the haloes. These can also be used to solve points 1 and 2 (above) but it still leaves the situation where a large part of the switcher will need to be used up when making cross-fades. It is operationally no better than before for points 3 and 4.

Solution 3: Use compensated video

It is in operation with still stores that the problem of the haloes occurs. Apart from this case linear keyers are fine and do an excellent job. It is not right to introduce another type of keyer or mode of keying just to handle the still store case. That is attacking the symptoms rather that curing the cause. It is good engineering practice to do the job properly and not to depend on downstream processing to compensate. It is the still store that is generating the problem, it is the cause, and so it is right that the solution is provided by that equipment.

The internal use of the key signal within the still store to make cutouts causes the edging problem downstream in the keyer. The edges of the cutout are mixes between video and background. Compensated video reshapes the mix profile so that it makes the correct result at the keyer output. The process is executed within the still store so the appearance of the compensated video prior to the final keying will show a slightly enlarged area but it is certainly good enough for a confidence preview. Operationally this works out very well when used with a standard linear keyer. This turns out to be an engineering serendipity where all problems are suddenly solved.

1) The stills may be previewed directly at the preview output. There is no need to look at the output of a keyer.

2) Since internal keying may now be used montages can now be assembled and displayed correctly at the output of the final keyer.

3) The problems of the halo were caused by unsuitable mixing of foreground and background. The solution that works to banish the halo also gives the required signal conditioning to create dissolve transitions at the output of the final keyer. This can now be achieved by the simple means of cross-fading both the keys and the compensated video within the still store.

4) Use is as simple as possible as the whole stills operation can be sequenced from the still store itself. There is no longer any need to switch between outputs as the correct sequence always appears out of the main output.

5) Transparent areas will be correctly shown throughout transitions.

CONCLUSIONS

The applications of still stores have greatly expanded since their introduction in the early 1980s. From its use as a slide scanner replacement to today's dual channel machines, offering cutouts, text and linear keys as well as full sized pictures, attention must always be paid to their interface into the production area. The connection of the outputs to the switcher requires special care as there are great benefits to be gained from getting it right.

At first it may seem that the technical and operational problems of introducing linear keys outweigh the advantages. The work arounds to lose the haloes and provide the flexibility of dissolve transitions involve the use of more equipment to the point that some switchers may use all three mix/effects banks. At the same time the simplicity of operation that had been developed for the use of whole picture stills, with all stills being sequenced through the one main output and available for checking through the preview, could be lost. Neither the use of whole pictures and separate downstream keyers nor the use of special compensated keyers offer a total solution and greatly increase the complexity of putting stills on-air... treating the symptoms not the cause.

The addition of further processing at the still store output to generate compensated video is the panacea for the problems identified with the use of linear keys. It treats the cause at its origin, in the still store. In this way the interface to the switcher can be resolved to a simple set-up requiring only the single main video and key output to be fed to the one keyer. The keyed result is halo free and can be faithfully cross-faded by still store operation. For on-air use it is only necessary to sequence the stills in the store, there is no need of any operation at the switcher end.

The switcher interface and its operation has been made as simple as possible through the application of extra processing in the output of the still store. The techniques involved are complex but digital technology has allowed them to be added as another building block to the machine. Not only is this of great assistance to the operation it is also another example of how digital systems can be extended towards providing exactly the required performance.

REFERENCE

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PLANNING SUCCESSFUL WIRELESS MICROPHONE INSTALLATIONS FOR TELEVISION

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This paper describes some of the more important considerations when planning Wireless Microphone installations for Television. Topics to be discussed range from small portable systems for remotes and location work to large transportable systems to permanent installations on sound stages. In addition, methods and procedures are described which have been found to produce optimum performance from Wireless Microphone systems. The results of these techniques are to maximize range with minimum interference, and to obtain superior audio quality for dramatic productions, documentaries, and news (ENG) for television broadcasting.

0 INTRODUCTION

Wireless microphones (RF or Radio Microphones) have found wide acceptance in professional audio applications at all levels. When properly used, Wireless Microphones work well and sound great. They solve many production sound recording problems, such as reducing the additional time required to light the sets to allow the use of overhead boom microphones. The ideal Wireless Microphone can be thought of as an invisible Microphone cable that neither adds or subtracts anything from the audio signal. There is a tendency to group the Microphone and the Transmitter together as one unit, when in fact there are a large number of choices of Microphones to use with most Transmitters, including hand-helds. In the future we will continue to see systems with smaller and more efficient transmitters, and more sensitive and selective receivers. This will occur as manufacturers increasingly incorporate surface mount techniques using large scale integration (LSI), and as the prices come down for the more exotic components, such as GaAsFETS. (Gallium Arsenide Field Effect Transistors).

1 EQUIPMENT SELECTION

Most manufacturers produce a range of models to fit most applications. Portable equipment includes miniature battery powered receivers and road cases usually referred to as Quad Boxes. Quad boxes hold up to four receivers, an antenna distribution system, and a battery pack. Battery packs for receivers and transmitters usually contain alkaline batteries. Quad boxes sometimes use rechargeable batteries. However, rechargeables are only effective when used on a regular basis. Portable receiver selection must consider ruggedness, because of the potential of abuse. The larger receivers usually give better performance, and are designed for rack mounting and AC power. Rack systems are available for either transportable or permanent installations. Hand-held transmitters usually are available with a choice of the popular Mic capsules. Most body-pack transmitters can be fitted with any lavaliere. There is even a plug-on transmitter with an XLR type connector that will accept any type of conventional microphone.

2 INSTALLATION CONSIDERATIONS

Permanent and semi-permanent installations require careful attention to AC power conditioning and proper grounding techniques. It is very easy to get into trouble with ground loops. In large installations audio distribution becomes a problem, and the solution is often the use of a so-called "Audio Snake". This is a multi-line cable that is available in multiples of up to fifty-two audio cables in one jacket. Antenna signal distribution also becomes unwieldy in large installations, unless antenna splitters or antenna distribution systems are used. Sometimes called multi-couplers, and available in groups of four to six outputs, these splitters can be daisy-chained to feed a large number of receivers from one antenna.

3 FREQUENCY ALLOCATION

The VHF frequencies used for Wireless Microphones are primarily 169 MHz through 216 MHz. As secondary users of these frequencies, the Federal Communications Commission has limited the Transmitter power to 50 Milliwatts, to preclude interference to the primary users, the Television Broadcasters.[1] The primary use for 169 MHz through 172 MHz is for Hydrological Research.

These frequencies are popular for itinerant applications. The UHF frequencies are 470 MHz through 904 MHz. The power limit for UHF is 250 Milliwatts. Wireless microphones have some limitations and require good frequency coordination to preclude interference problems.

4 EFFECTIVE RANGE

Experience indicates that the range of UHF compared to VHF is something on the order of two or three times greater, even taking into account the increased path losses for UHF. Very few of the UHF transmitters are operated at the legal maximum power level because of the severely reduced battery life. It is very difficult to state an effective range figure for Wireless Microphone systems because range is dependent on so many factors. I have personally encountered effective range of over a mile with VHF on the one extreme and barely three feet on the other. The reality of dependable range is on the order of 300 to 500 feet in average conditions.

5 SIGNAL ABSORPTION

The signal transmitted by a wireless microphone is absorbed by nearby objects, including people. Transmitter power measured under laboratory conditions may be 50 mw. The power measured from a transmitter worn by a performer is another matter entirely. With the transmitter held about 55 cm away from the body, the power can be down to about 25 mw. With the transmitter held about 20 cm away, the power can be down to about 2 mw. [2]

6 RECEIVER SATURATION

At times, wireless microphones appear to have no range at all for no apparent reason. Often, this is caused by receivers with poor design or with wide bandwidth RF front ends being desensitized by RF energy on nearby frequencies.

7 CO-CHANNEL INTERFERENCE

Co-channel interference usually occurs due to poor or non-existent frequency coordination. This is usually inter-modulation distortion where the harmonic of one frequency mixes with a second frequency (or with another harmonic of a second frequency) to produce a frequency at or near the assigned frequency of the receiver. The frequency relationships which cause the problems are often obscure.

8 DIRECT INTERFERENCE

This usually occurs in situations where an additional wireless is added to an existing set of units without proper frequency coordination. For instance, news crews (ENG) often interfere with each other when they all arrive simultaneously at an event because of the same lack of coordination. This can render all units unusable with no range and audible interference. The noise is the difference between the two carrier frequencies. It is called heterodyne and sounds like canaries chirping. The only way to restore normal operation is to turn off the duplicated frequency transmitter.

9 NEON

Especially when brought in by the production company, Neon lighting is always an interference source. This is because it is usually a temporary installation which radiates broad band noise across the entire radio Mic spectrum, including UHF. Using the right size ballast on the Neon and sealing the connections with anti-corona silicone compound are the only techniques that seem to help.

10 SELECTIVITY

Unfortunately, it is expensive to produce a receiver with good selectivity (the ability of a receiver to ignore nearby frequencies.) Good selectivity is often the difference between popular and unpopular brands or models.

11 MULTI-CHANNEL RECEIVERS

Several manufacturers produce multi-channel receivers. This design has one advantage, which is to allow an operator to switch to a new frequency when he finds an existing frequency to be unusable. The main design disadvantage is that it cannot be made to exhibit good selectivity.

12 COMPANDERS

The use of audio processing (COMPANDERS) has become standard practice in modern wireless systems. All presently available audio processed wireless microphones use some form of cooperative compressor/expander circuitry. The term "cooperative" is used because signal processing is applied at both ends (Transmitter and Receiver) of the audio link. In properly designed systems, dynamic range of 100 dB or better is common. An additional and often overlooked benefit of processing systems is an improvement in effective working range.

13 MULTI-PATH CANCELLATION

The radiated energy from a transmitter is radiated in all directions. The signal picked up by the receiver will include the direct signal and any signals reflected by nearby objects, such as light and grip stands, reflective walls, and metallic objects. Due to the longer travel distance for the reflected signal, it is delayed slightly and will partially or completely cancel when mixed with the direct signal. When the signal level drops below the squelch threshold, the receiver audio is muted momentarily and is usually accompanied by a "squelch tail."

13.1 MULTI-PATH THRESHOLD

Multi-path threshold refers to the signal strength where the squelch circuit cuts off the audio. This is the objectionable noise associated with multi-path dropouts or "hits" as they are called. There are several ways to reduce this problem. By maintaining a high signal strength at the receiver, the dropouts will not go below the squelch threshold. Keeping fresh batteries in the transmitter and keeping the receiver as close as possible to the transmitter help minimize this problem.

14 DIVERSITY RECEIVERS

There are several equipment designs that help reduce dropouts. The most effective are antenna diversity and receiver diversity. The antenna diversity design combines three separate antennas in an isolation amplifier and three way summing network that greatly reduces the dropout problem. However, it is mathematically possible to have dropouts occur with this design. The advantage of antenna diversity is that it is much less costly to implement. The disadvantage in a portable operation is that three antennas must be connected and separated, preferably by several wavelengths (the distance a radio wave will travel in one cycle.) Receiver diversity utilizes dual receivers and an electronic switching circuit that switches the audio output to the receiver with the best signal strength. Because the null point in a multi-path dropout is usually just a few centimeters long, separating the antennas for both receivers by more than a quarter wavelength is all that is required to gain much improvement over non-diversity. The advantage of receiver diversity is that just two antennas are required. However, the receiver design is much more expensive. Receiver diversity is more popular than antenna diversity for production work not only because of its superior function, but because it also reduces the effort required in setting up and moving antennas from shot to shot.

14.1 DIVERSITY ANTENNA SEPARATION

Separating diversity receiver antennas is a confusing issue. Some manufacturers permanently attach both antennas just 15 cm apart. While some improvement is realized, it is generally agreed that the antennas must be separated by at least one quarter wavelength or more to get the full benefit. There appears to be no additional improvement gained with separations greater than two meters.

15 TRANSMITTER OUTPUT POWER VERSUS BATTERY VOLTAGE

The transmitter output power is directly proportional to battery voltage in most designs. This is done in an effort to increase efficiency. However, this means that after six or eight hours of operation, the battery voltage is down and the output power is down accordingly. The tendency of many operators of wireless mics is to stretch the life of the batteries. This is not a good practice. Batteries are not expensive compared to the cost of lost production time or unusable dialogue. Most production mixers insist that the batteries be changed after three or four hours. Sennheiser is one manufacturer that regulates the voltage in the output stage of their transmitters so that the power is constant over the life of the batteries.

16 TRANSMITTER ANTENNAS

Most designs include a separate antenna which should be carefully separated from the Mic cable. A frayed antenna is a noise source and should be checked regularly. Some operators test the antenna timidly. The best advice is to test by pulling hard on the antenna. If the antenna wire is weak and about to fail, it is desirable to have it fail while testing rather than when it is in use. Several designs do not have a separate antenna at all. The shield on the Mic cable is employed as the antenna. This design appears to be quite effective because it eliminates one source of trouble. Also, the majority of the Mic cable is in front of the performer where the receiver is usually located, thus reducing the effects of signal absorption through the body.

17 RECEIVER ANTENNAS

Most wireless systems are supplied with a quarter wave "whip" antenna, very often a helical or "rubber duck" design with the portable equipment. This is the most inefficient antenna of all, especially when it is attached to a transmission line. Technically it is not even an antenna. A much more effective antenna is a half wave "dipole." See Figure 1.

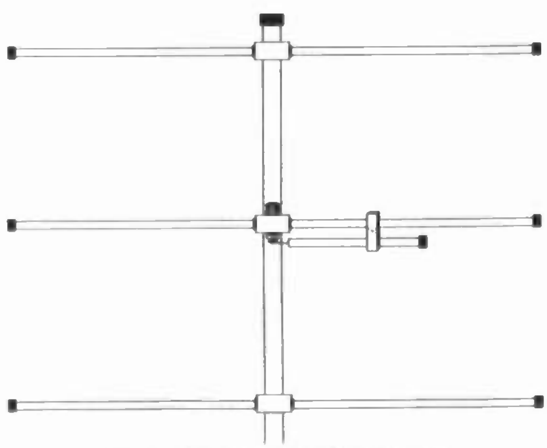
Figure 1

Vega Model 123 Dipole Antenna

A dipole antenna provides 3 dB more gain than a whip, and when operated with vertical (straight up) polarization, it is an omni-directional antenna as well. For best performance, both the receiving and transmitting antennas should be positioned with the same polarization. Several manufacturers make coaxial dipoles, where the transmission line is connected through the center of the bottom element. Another popular design is the end-fed half wave. These designs provide more uniform reception from all directions and lend themselves to easier mounting. The best range performance will be obtained with a "hi gain" antenna of which there are several designs, including Log Periodic, Cubical Quad, Collinear, and Yagi. The most popular design for production work is a "Yagi". (see Figure 2)

Depending on the number of elements in the design, the gain can be from 6 dB to over 11 dB. The Yagi is a directional antenna which must be pointed at the transmitter. The directional characteristics can be used to advantage if an interference source is identified.

Figure 2



Three Element Yagi Antenna

18 SIGNAL TRANSMISSION

An area of great confusion exists regarding transmission lines, or antenna cables, for wireless mics. Most users prefer to keep their receivers near the recording channel, out of harm's way, and run out an antenna on a long cable when the need for extra range arises. This technique yields limited success due to the large losses in the commonly used cable. RG-58/U coaxial cable has a loss of 4.5 dB for 22 meters at 180 MHz (see Figure 3). RG-8/U and RG-11/U have correspondingly less loss. However, they do not find widespread use in the television industry because they are very stiff and difficult to use. The best solution is to put the receivers on the end of a long audio snake cable and attach the antenna with a short transmission line.

Coaxial cable loss for VHF @ 180 Mhz

Length (m) of Coaxial Cable	Loss (dB) of Standard RG-58/U	Loss (dB) of Standard RG-8/U	Loss (dB) of Polyfoam-Type RG 8/U
15.15	3.00	1.80	1.00
22.72	4.50	2.60	1.50
30.30	6.00	3.50	2.00
37.87	7.50	4.40	2.50
45.45	9.00	5.30	3.00
53.03	10.50	6.10	3.50
60.60	12.00	7.00	4.00
75.75		8.80	5.00
90.90		10.50	6.00
121.21			8.00

Figure 3

The next best choice is air; that is, use the high gain antenna close to the receivers and the receivers close to the recording channel. The third choice should be the long transmission line. The use of antenna amplifiers has proven to be unsuccessful due to noise problems caused by inter-modulation distortion. One antenna feeding four receivers utilizing an active splitter (multi coupler) incorporated in a road case (Quad Box) is very popular. Most manufacturers offer such systems. The use of passive splitters is not recommended due to the insertion loss. The loss of any signal strength can not be tolerated.

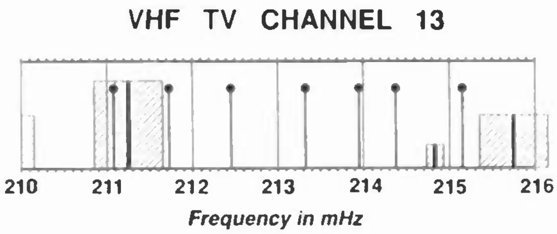
19 FREQUENCY SELECTION

As stated earlier, the most popular VHF frequencies are from 169 MHz to 216 MHz. The 169 MHz through 171 MHz band is referred to as the "traveling band" because these frequencies can be used almost anywhere with little fear of interference. The primary application for this band in the United States is for hydrological research. It is easy to find four compatible frequencies in this band. Because of its popularity, the traveling band has become a problem with newscasters. When a large group of ENG crews get together at an event, they interfere with each other. 174 MHz through 216 MHz represents the television channels 7 through 13. Operators have had good success operating on the unused TV channels in their area. In addition, we have found good success using what are referred to as "slot" frequencies in the active TV channels. It is possible to find four and sometimes five compatible frequencies on either side of the picture and sound carriers. (See Figure 4)

20 VHF/UHF

The major advantages of UHF over VHF frequencies are the large number of frequencies available and some increase in range. Stage shows regularly use twenty, thirty, or more wireless at the same time. Attempting this with VHF would be very difficult.

Figure 4



Usable Slot Frequencies
in Unshaded Area

The major disadvantage of UHF is that the equipment is much more costly due to the larger number of components and their closer tolerances. In the upper UHF band, around 900 MHz, there is some immunity to impulse noise, such as high energy ignition systems in race cars.

21 ADVANCING TECHNOLOGY

Wireless microphone performance has improved substantially in the last few years. Today's miniature receivers work as well as yesterday's giants. Most of the improvements in wireless microphone systems are intended to reduce interference and improve audio quality. The ultimate goal is for the users to have a system that is transparent in terms of audio quality. This is one area where it really makes sense to go with the newest designs.

22 TRANSMITTER PLACEMENT

Resist the temptation to just slip the transmitter in an actor's pocket, clip on the lavaliere, and think that you are ready. Unless the transmitter is placed so that there is no danger of it falling out, you are going to have problems. At least one company makes an excellent belt and pouch kit for placing the transmitter around the waist, usually in the small of the back. Many of the transmitters have good belt clips, and most are removable when necessary. A number of other ideas work well also. When the actors are dressed in military or police costumes, the transmitter can be placed in an ammunition clip holder on the belt. Actresses with form-fitting costumes present genuine challenges for transmitter placement. One idea is to put the transmitter in a purse when available. Sometimes, it is necessary to place the transmitter on the ankle of an individual with tight-fitting clothes. A product that works very well for this application is called "Coban," a self-adherent wrap elastic bandage. Manufactured by 3M, it is available from most medical supply stores.

23 LAVALIERE MICS

Most microphone manufacturers make lavaliere type microphones, and they all work well. Certain models have become popular in Hollywood for a number of reasons. Lavaliere is a name that was applied to chest worn microphones when they were very large and were supported by a neck strap. With the latest designs, the lavalieres are so small that they are extremely easy to conceal.

24 END ADDRESS/SIDE ADDRESS

Most lavalieres are designed as end address, tubular in shape with the sound opening at the end. Typically, this design is more difficult to make noiseless when concealed than the side address style. When lavaliere mics are used out in the open, clothing noise is rarely a problem. The side address style lavalieres lend themselves to a wide assortment of mounting accessories for concealment.

25 OMNI/CARDIOID

Normally, lavalieres used for television work are omni-directional designs. The cardioid design is used for sound reinforcement applications where feedback is a problem. When using cardioid designs for television work, the problem is that actors tend to be more animated than public speakers, make a lot of head turns, and tend to turn out of the pick-up pattern of the Mic.

26 OPEN/ISOLATED

One characteristic of lavalieres that is not well understood is that some designs tend to provide more isolation than others. Lavalieres that are open and do not tend to isolate can be used to pick up voices from a distance of a meter or so and make good "plant" or concealment mics. They are good for concealing on a desk or table and picking up the dialogue around the table. The Sennheiser MKE-2 is an example of an excellent open type lavaliere. Microphones that tend to isolate and only pick up voices in close proximity are easier to use with wireless microphones, especially when several units are mixed down to one track. Using wireless microphones, if all the Mic inputs are left open, causes phasing problems when a voice is picked up by two or more microphones at different distances from the speaker. Lavalieres that tend to isolate reduce this problem and allow the mixer some latitude in this regard. This is helpful when the mixer doesn't know the dialogue cues. With the more open lavalieres, the mixer is forced to "cue" the mics on and off to avoid the phasing problems and to reduce the noise buildup. This is a very important consideration for wireless applications.

An example of a lavaliere that provides good isolation is the Sony ECM-55.

27 CLOTHING NOISE

Clothing noise is probably the most pervasive problem caused by the use of lavalieres and wireless microphones. This is the main reason that these mics are difficult to use in quiet locations. Heavy background noise tends to mask much of the clothing noise. The most popular lavalieres for film work are side address models, the Tram and the Millimic (half the size of the Tram.) These lavalieres are very easy to quiet when concealed. This is partially due to a comprehensive assortment of very effective mounting accessories. These mics lie mid range as far as openness is concerned, so they do require some cueing when multiple mics are utilized. Regardless of which microphone is used, some ideas follow that help to minimize clothing noise. One idea is to request, when possible, that performers refrain from wearing polyester or starched shirts or silk ties. An idea that is helpful for reducing clothing noise when two items of clothing are brushing together, such as a jacket and vest for example, is to stick them together with double face hair-piece tape. Available from beauty supply stores, this is a very thin 2.5 cm by 7.5 cm clear tape which is also quite useful for attaching tubular shaped lavalieres. Another very handy product is called "Moleskin." Manufactured by the Dr. Scholl's Company, it is a 7.5 cm by 10 cm multipurpose sheet of adhesive backed, thick flannel cloth which can be used in attaching and quieting lavalieres. Not only is Moleskin convenient as a backing for the microphone, but it can be used for mounting and covering the lavaliere also.

28 CONCLUSION

This paper has investigated the limitations and challenges of wireless microphones as they are currently used in the Television Industry. Some of the procedures and techniques that allow production mixers to obtain consistently good audio quality using this technology are discussed.

29 ACKNOWLEDGMENTS

The author wishes to thank several of his colleagues at Audio Services Corporation for their encouragement and helpful discussions. These include Richard H. Topham, Jr., and Michael Riner.

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AN OVERVIEW OF ONE STATION'S EXPERIENCES WITH ROBOTIC CAMERAS

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Implementing robotic camera systems at the local station level requires engineering management to coordinate all aspects of the project with the rest of the stations' departments.

Robotic systems in general are highly reliable, flexible and operationally functional, but due to the inherent differences between one person-one camera operation and the operation of robotic systems, some changes will need to be considered when the decision is made to purchase a robotic camera system. This paper will describe those differences and will discuss the economic impact and justification of the purchase and the decisions made at one local television station to make the conversion to robotic camera operation successful.

Successful conversion to robotic studio cameras requires consideration of a variety of issues beyond the traditional engineering concerns of reliability, design and factory support.

Man versus machine comparisons are inevitable; staff sensitivities regarding automation systems in general must be addressed; and the impact of a six figure capital purchase on the bottom line will be closely scrutinized by management and must be justified.

Still, with careful planning, robotics can be justified and, in the case of KDFW, can even exceed initial expectations.

The Decision To Purchase

As with any major capital project, basic questions need to be answered prior to purchasing. First and foremost, would the performance of robotics have any effect, either positive or negative, on our studio product -- consisting mostly of live news. We anticipated -- correctly as it turns out -- that the robotics would be an improvement over manned cameras in certain situations, and not as good in others. For example, precise repeatability is easily achieved with robotics, but extreme close up shots are difficult to frame and maintain.

Detailed discussions with the News and Production departments led to the adoption of standardized studio procedures which minimized the limitations of the robotics while at the same time not adversely impacting the on-air look. Whenever possible, we pre-record extreme close ups on still store. By adopting this simple solution, a potential problem was resolved before it became a problem. At KDFW it was critical to our success with robotics that issues such as this were dealt with before, rather than after, we implemented the system.

On balance, then, it was decided that robotics would have some positive and minimal negative impact on our studio on-air look. The second question dealt with economics. Since we had determined that there would be no dramatic impact one way or the other on our on-air look, the decision to purchase rested most heavily on the cost versus the benefits of robotics. A not-so-subtle side issue of the economic question was the effect on department morale, should the acquisition of robotics result in staff layoffs. The cost analysis side of the economic equation was easily calculated by determining our needs.

We have three full sized studio cameras in the main news studio and one additional camera in the newsroom for a headline and weather alert set.

Although cameras one and three needed full mobility, to cover weather and sports, camera two and the newsroom camera needed only pan, tilt, zoom and focus robotics. We were able to lower the initial purchase price of the system by tailoring it to our specific needs, but can still upgrade cameras two and four to full mobility in the future if needed. Using this building block approach meant that the project came in significantly under budget. Like all good engineering managers, I immediately came up with a way to spend the left over money. We decided to expand the system by adding robotics to a camera at our Ft. Worth news studio, to allow control of that camera, thirty miles away, from our Dallas studio.

Savings Analysis

With the expense analyzed, we then examined the cost savings associated with robotics, which proved more difficult.

Automation systems of all sorts have traditionally been justified by their ability to reduce staff and "pay for themselves". This sets up an immediate and significant catch-22: The very people charged with successfully installing and operating robotic camera-system feel that they have the most to lose from that success. At KDFW, this potential conflict was addressed as soon as the decision to purchase robotics was made.

It was decided and communicated to the engineering staff that upon implementation of the robotics, the extra studio camera people would be re-trained in other areas. This solution had the dual benefit of providing demonstrable cost savings to the station in the form of reduced overtime to these areas, while at the same time maintaining the existing staffing levels and relieving any fears regarding layoffs.

Of course, each station will be different, but, we found that careful consideration and communication to the staff of the impact of robotics on manpower is essential to a hassle-free switch from manual to robotic studio cameras.

The final economic question is where does the project fit into existing station priorities. At KDFW capital money originally budgeted for studio cameras to replace our aging RCA TK-47's was diverted to the robotic project. The rationale for this decision was based partly on simple economics: The robotics would provide quantifiable cost reductions in operations. The decision was also based on our desire to take a "wait and see" attitude with regard to developments taking place with studio cameras for NTSC/wide scan NTSC and HDTV. The budgeting process will vary by station, but the point remains that now, more than ever before, engineering management must make decisions and set priorities with a view toward all facets of current station operation, as well as future requirements.

Installation

Once the decision to purchase had been

made, the installation and operational details had to be worked out. During this phase of the project the engineering department kept all other departments informed of our progress. Meetings were held with anchors, stage managers, robotics operators -- in short, anyone who would be affected by the new system.
Early on, it was decided that the robotic operations console would be co-located with the cameras in the main news studio. This was done so the operator could physically view the cameras in relation to other objects in the studio, namely, people.

An additional benefit was gained by locating the camera RCU's at the robotic console, enabling the operator to easily chart and setup the cameras from the studio. Finally, although we felt the technology was sufficiently redundant to prevent a total catastrophic failure of the system, the in-studio operator could quite easily and quickly disengage one of the robotic heads and manually operate a camera if that ever occurred.

The backup controller was placed in master control to allow robotics operation by the master control operator during overnight or weekend periods when single camera news headlines might be required.

The robotic interconnect is serial based, allowing for lengthy runs between the two operating stations, and the cameras themselves. For convenience sake, the power supplies used to charge the pedestal DC drive motor batteries were all mounted in the studio. All cables were fed through nylon webbing to the camera heads and strain reliefed at the pedestal.

One minor problem, involving interference to our VHF wireless intercom system by a pedestal CPU clock was resolved by the simple expedient of relocating the intercom base station to a far corner of the studio.

Training

Operation of camera robotics is fairly intuitive, and was made more so by the fact the operators could see the cameras respond to commands since they were located in the studio. All of our operators became quite comfortable "driving" the cameras in less than a week. During training, we did discover some differences between manual and robotic operation. Primary among these differences was reaction time from the moment a director called for a shot and the shot was ready.

Since the camera person no longer got instantaneous sensory feedback as he or she did when manually pointing the camera reaction time increased. With robotics, the shot is accomplished by manipulating the joysticks on the robotic control console, or, more simply, using the icon based bit pad. The bit pad greatly simplifies camera positioning. We have found that basic shot acquisition is best obtained using it and final touch up, if needed, is done with the joysticks. The directors' quickly became acclimated to the difference in reaction time and adjusted for it easily.

All training was done between regular newscasts with full crew rehearsals. During training we did have a few non-critical bumps, but the robotics are very well protected against collisions and we put up warning signs to alert people walking by or standing near the cameras that they might move unexpectedly. If anything simplified our cut-over to robotics, it was the decision to rehearse complete thirty minute newscasts, with full crew and talent. We taped the rehearsals, with P.L. audio recorded as well, and reviewed each rehearsal afterwards. The cut-over itself proved to be anticlimactic. We literally were operating manually one day, and with robotics the next.

Operating SANS Operators

After cutover to full robotic operation about the only serious problem we encountered but did not anticipate was the operation of the robotics during studio interviews.

Robotic operation lends itself ideally to a fixed format: Tightly scripted, repetitive shots. Exactly what we are used to during daily newscasts. " In studio" interviews, however, are a totally different environment. Little or no script, out of station guests, rather than trained talent, and rarely, if ever, a fixed format. A single operator using joysticks simply cannot do as good a job in this situation as one operator per camera can. Still, with the cooperation of news and production we quickly found a solution. Interviews are always scheduled immediately out of a break, which gives the robotic operator two minutes of uninterrupted time to re-position the cameras to the interview area of the news set. Cover shots are entered into the robotic memory and directors have been instructed to keep at least one cover shot ready at all times. Talent is alert to the differences between manual and robotic operation and have adapted accordingly. With these slight modifications, our initial problems with interviews have disappeared.

Remote Operation

Dallas and Ft. Worth are known locally as the metroplex, but each city is very locally oriented and neither wants to be overwhelmed by the other. We had wanted for some time to increase our exposure in Ft. Worth and had investigated a dual location, Huntley - Brinkley approach to our prime newscasts.

We did some tests with an anchor in front of a couple of locked cameras at our Ft. Worth news bureau and intercut them with the anchors in our Dallas studio. The different "look" was immediately apparent and unpleasant. The static shots from Ft. Worth contrasted poorly against the fluidity of the robotic moves in Dallas.

Stationing a full-time camera person at Ft. Worth to correct this problem was out of the question, given the current reception requests for additional staff receive in the General Managers office. The solution was found in our new robotic camera system.

The system has provisions for standard 9600 baud communications over a data line to enable control of one or more remote camera robotics. Utilizing this feature allowed us to operate the camera in our Ft. Worth studio from the operators position in Dallas.

We use a second locked down camera in our Ft. Worth studio for an over the shoulder shot. The technical director in our Dallas control room switches between the two cameras via a second data line controlling a small routing switcher in Ft. Worth. With only one microwave link from Ft. Worth to Dallas, the shot memory and repeatability of the robotics became critical since the robotics operator was not always able to preview the upcoming camera shot. Fortunately, the system is excellent in this respect and our Ft. Worth studio is now seamlessly integrated into all of our newscasts.

Expandability

The system is configured to control eight cameras, which originally seemed excessive to us. But after only six months of operation we were already up to five cameras and have been discussing the possibility of adding robotics to the other camera in Ft. Worth, which would bring the total to six. Initially we had a concern about the number of cameras a single operator could reliably and accurately control.

The operators themselves thought that four was very nearly the upper limit for error free operation.

Experience has taught us, however, that since the camera shots are scripted and therefore very predictable, one operator running five cameras has proven to be if not easy, certainly manageable. Further, there has been no concern on the part of the operators, with the idea of operating a sixth robotic camera now that they have gained some experience.

<u>Summary</u>

The current generation of camera robotics equipment is capable of flawless operation at the local station level. The systems offer expandability, flexibility and ease of operation. Innovative solutions to issues such as remote studios and news bureaus are also possible with robotics and offer further justification for investing in these systems. Application of the technology, however, requires diligence in planning by engineering management.

The fact is, there are significant differences between human operators and robotics, and everyone, from talent to management to staff personnel needs to be made aware of them so differences are not mistaken for problems.

DESIGNING CBS-TELEVISION'S FIRST SERIAL DIGITAL, MULTIPLEXED, VIDEOTAPE EDITING ROOM

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ABSTRACT

CBS Television City recently completed the design and construction of a multiplexed serial digital, video tape editing room. The room, Edit 12, is now in use by CBS's highly rated daytime drama **THE YOUNG AND THE RESTLESS.** In the areas of system design, construction techniques, operational features, maintenance requirements, and overall value, Edit 12 represents a major departure from the traditional analog edit room. Edit 12 is also unique in the fact that it is among the first to incorporate program path multiplexing of audio and video. As the construction of Edit 12 concluded and system checkout began, the tremendous wide-ranging advantages of multiplexed serial digital became very clear. Multiplexed digital production facilities are less expensive to construct, easier to maintain, and more efficient to operate than either analog, parallel or non-multiplexed systems. This paper will discuss problems, solutions, and observations made during the design, construction and subsequent successful use of this video tape editing room. System design, equipment selection, construction technique, and overall plant integration will also be covered.

INTRODUCTION

Initially, the design, construction, and implementation of a serial digital, multiplexed edit room may seem expensive and risky, but the experience gained by CBS Television City during the construction and operation of the network's first serial digital video tape editing system, Edit 12, confirms that this is not the case. The multiplexing of digital audio and digital video allows a significant reduction in hardware and wiring while at the same time simplifying the entire edit system. Multiplexing also allows a single level digital router to route both audio and video simultaneously, thus eliminating an entire routing switcher level. Analog plant integration has always been a concern when contemplating a digital facility. The serial production switchers now available have auto-timed analog inputs allowing simplified analog plant integration.

In essence, this paper is a guide to successfully building a serial D2, multiplexed A/V, video tape editing system. It is the author's intention that this information will prove helpful to other engineers and/or managers who are in the initial stages of designing their own digital post production facility.

CBS TELEVISION CITY'S EDIT 12

Facility Design

The space allocated by Television City for the construction of new digital edit rooms measured 37-feet long by 28-feet wide. The space was then divided into 2 separate edit rooms at 342 square feet each. The remaining space was used as a common equipment room servicing both Edit 12 and a future edit room. The ceiling of the allocated space had been acoustically designed for a previous technical facility and was a determining factor in the Edit 12 layout. By taking advantage of the existing ceiling's corner rise, Edit 12 required a minimum of sound defining devices while still providing a good stereo image. Sound isolation between the three rooms was accomplished by the use of high loss frame walls with neoprene gaskets. Figure 1 shows the Edit 12 facility layout.

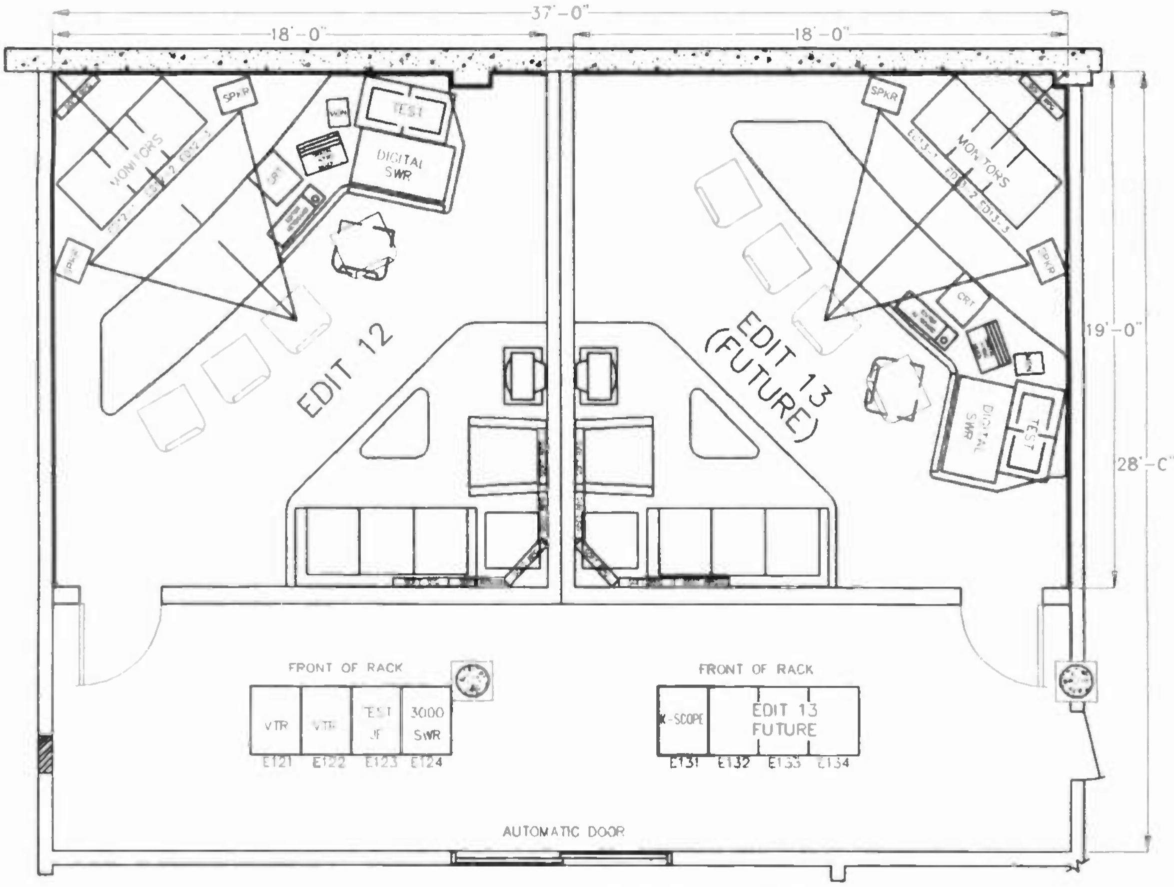


FIGURE 1: EDIT 12 FACILITY LAYOUT

Typical System Diagram

The simplified block diagram of a serial D2, multiplexed editing system is shown in figure 2a. For simplicity, the diagram illustrates a 2 machine editing system, any additional D2 VTR's would connect in the same configuration. The digital outputs of each D2 VTR are multiplexed together by an internal VTR board. After multiplexing, each VTR output is then sent via coax to an external demultiplexer. The "demux" splits the digital bit stream apart, sending audio to the audio mixer and video to the video switcher. After processing by the audio mixer, program audio is then sent to be externally multiplexed with the program video. After "muxing", the digital bit stream is then distributed to the multiplexed input of each VTR. The program path length of the audio mixer is less than 1 line of video. This delay causes no problem when muxing or demuxing.

After reading the previous paragraph and looking at figure 2a, you may be asking yourself a question, "If the A/V bit stream is just going to be demuxed at the switcher input, why use the VTR's internal mux board at all ?". The answer can be seen in figure 2b.

Figure 2b is the same as figure 2a except for the serial routing switcher that was inserted between the multiplexed I/O ports of the VTRs and the multiplexed I/O ports of the edit room. Muxing and demuxing allows a single level serial router to route 4 channel audio, video and VITC time code, all on a single coax. Television City is presently installing a new serial routing switcher. When this router comes on line, Edit 12 will be connected as shown in diagram 2b.

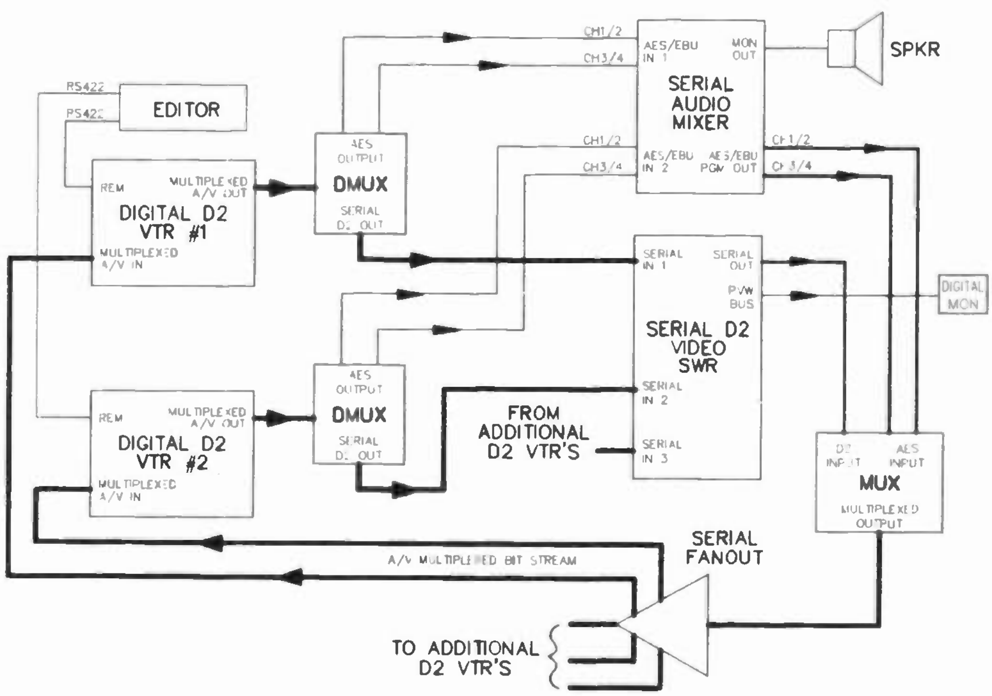


FIGURE 2A: TYPICAL BLOCK DIAGRAM MULTIPLEXED EDIT SYSTEM

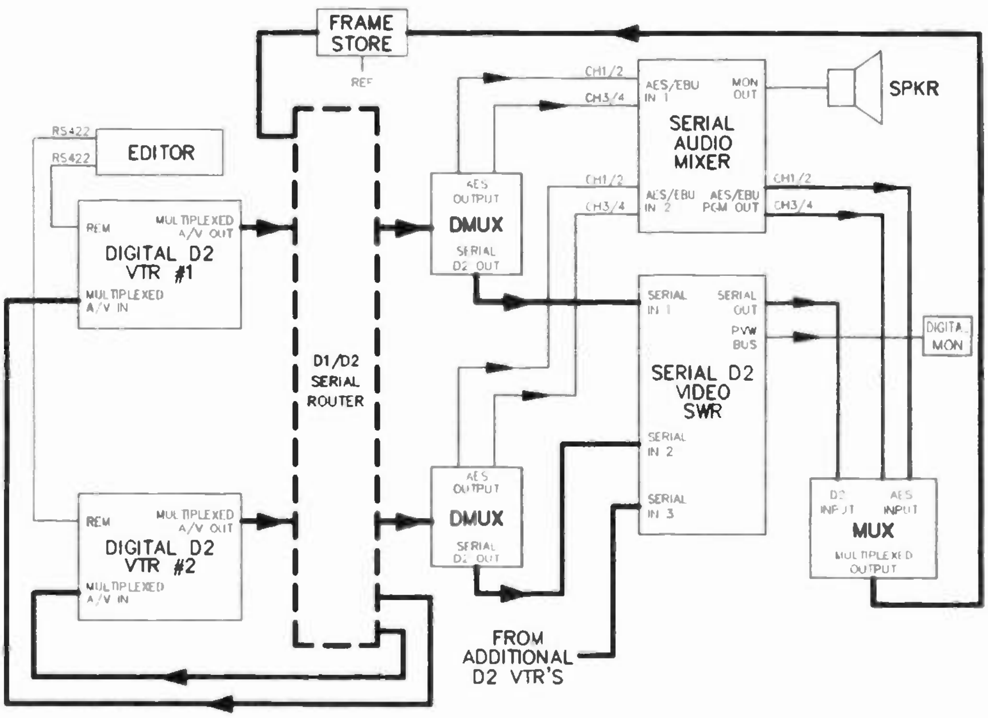


FIGURE 2B: TYPICAL BLOCK DIAGRAM MULTIPLEXED SERIAL ROUTER

Audio breakaway is a concern when dealing with multiplexed serial digital routers. This problem can be solved during an edit session by using the audio mixer to perform the breakaway. In a dubbing situation, other methods could be used to achieve the desired breakaway. In either case, breakaway audio does not present a problem when routing multiplexed A/V.

The editor controls the VTRs using the standard RS 422 control cables. The traditional external preview selector is not needed in a serial digital editing system. The selector has been replaced by the video switcher's internal Aux bus. The Aux bus, in communication with the editor controller, provides a high quality digital preview.

EDIT 12 SYSTEM DESIGN

The simplified audio/video block diagram of Edit 12 can be seen in figure 3. This block diagram is similar to the previous diagrams but also includes analog sources, serial test stations, sync generators, and a monitor router. These additions are discussed in the following paragraphs.

Analog Sources

Analog integration with Edit 12 was accomplished by the use of optional, auto-timed, analog inputs. These inputs are available on both the serial D2 video switcher and the digital audio mixer. As seen in figure 3, the analog VTRs and the analog DVE are directly connected to the analog inputs of Edit 12's audio mixer and video switcher.

Because of a rate conversion problem, the DVE is not digitally integrated with Edit 12. Since the DVE is parallel D1 and Edit 12 is serial D2, high quality rate conversion equipment would be required to

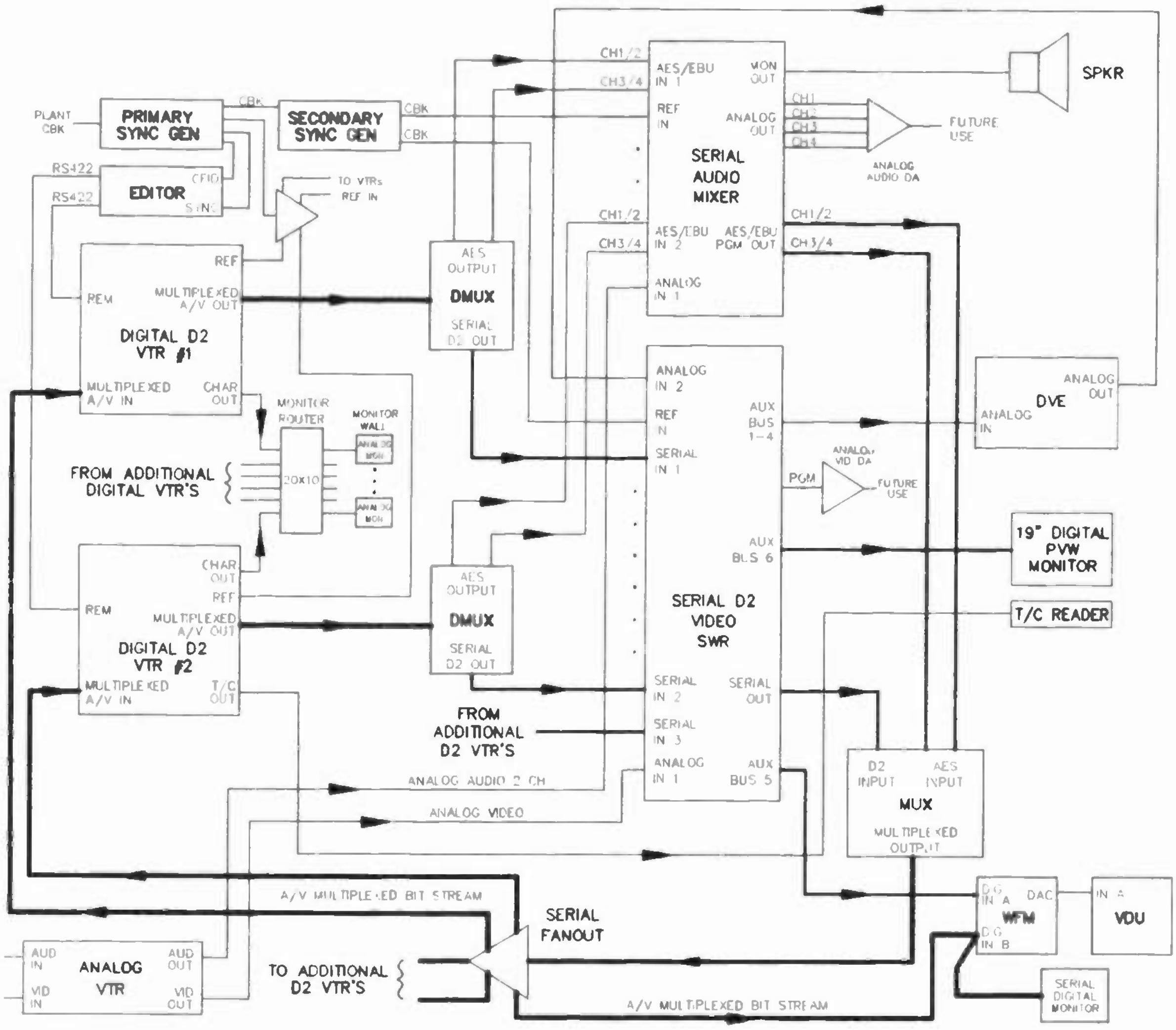


FIGURE 3: EDIT 12 SIMPLIFIED BLOCK DIAGRAM

make this connection. At the time of Edit 12's construction, the cost of reliable conversion equipment did not justify the resultant increase in video quality. As soon as it is cost effective, the Edit 12 DVE will be digitally installed.

Serial Test Stations

Edit 12 includes two serial digital test stations. Both stations consist of a vectorscope, a waveform monitor, and a 13" color picture monitor.

The first station, located in the equipment racks, is fed digitally with multiplexed program and switcher Aux bus 5. The waveform monitor has an internal, 10 bit, D to A converter. The D to A converter output is used to feed the analog vectorscope. The second serial digital test station is configured in the same manner and is located inside the edit room.

The Edit 12 test stations have two main purposes. The first is to check the digital integrity of the serial bit path, the second is to check picture content, video levels, SCH phase, and overall picture quality.

The digital integrity of the bit stream can be checked using the waveform monitor's on screen displays. Indicators are given for error data handling and audio channel validity. A digitized picture of the serial EYE pattern can also be displayed. An equalization button on the waveform monitor allows observation of digital signals in the equalized or unequalized modes.

Notice that the test stations are not used for input timing adjustments. This is very different from a traditional analog edit room where the test station is sometimes called a timing station. Except for the optional RGB chroma key inputs, all Edit 12 switcher inputs are auto-timed. The capture range of the auto-timing feature is +/-17.5 microseconds for both digital and analog inputs. For coax cables with propagation factors of 66%, 1 microsecond equals 648 feet. Multiplying +/-17.5 x 648 gives a possible coax cable length of +/- 11,340 feet. The video switcher also has an output delay of 1.18 lines, or 75 microseconds as shown in figure 4. Due to the auto-timing inputs and the fixed output delay, any source selected on Aux bus 5 will appear timed. If for some reason actual input timing needs to be checked, serial digital jacks are available for this purpose. At some future date, a software upgrade for the video switcher may allow the auto-timing inputs to be switched off. Aux bus 5 could then be used for timing purposes.

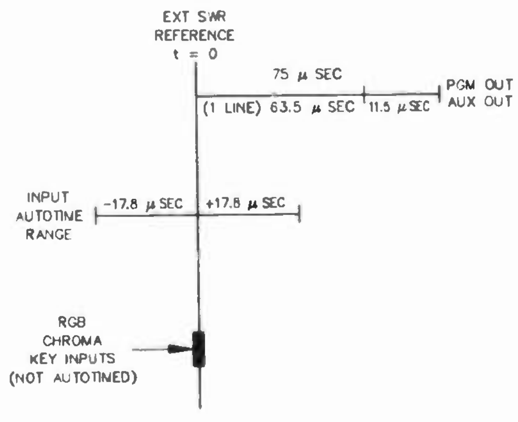


FIGURE 4:

Due to the 75 microsecond delay, 11.5 microseconds more than a line, the test station and all picture monitors on the output side of the Edit 12's switcher are not referenced externally. If they were, the waveform monitor's back porch clamp would be performed 11.5 microseconds into a line instead of during the back porch. This would appear as shifted video on the display. The picture monitors would also exhibit this shift. Edit 12 has external reference connected to both test stations, but it is normally not selected.

It is important to note that due to the video switcher's 75 microsecond program path delay, external re-entry can not be accomplished without a frame store device. Remember, the timing window of the switcher is +/-17.5 microseconds, video that is 75 microseconds late will not auto-time.

The SCH phase of any new switcher sources should be checked. The Edit 12 video switcher attempts to correct all SCH phase errors. This is a useful switcher feature but care should be taken in it's use. SCH phase by definition, shifts 180 degrees between odd and even fields. This means if the SCH of a switcher source is more than 90 degrees off in the odd direction, when the field is really even, the switcher will pick the odd phase and make an improper correction. If this were to happen, color frame errors would surface during matchframe editing.

The vectorscope provided in Edit 12 is used to check absolute and relative SCH phase. When checking relative SCH, the vectorscope should be externally referenced and the signal under test should be patched into the digital test station. Due to the auto timing feature, Aux bus 5 can not be used as a source selector when performing SCH measurements.

Sync Generators

Edit 12 has 2 sync generators, a master generator for the VTRs and a secondary generator for the audio and video switchers. The master sync generator is locked to plant color black and feeds each VTR as well as the secondary sync generator. The master generator also feeds the editor controller with sync and color frame ID pulses. The secondary sync generator feeds the audio switcher, the video switcher, and the DVE. This configuration allows the video switcher to be independently adjustable from the source inputs. Unlike most analog production switchers, the Edit 12 video switcher does not generate color black for external use. The color black it does generate is used only for program output.

A switcher dedicated sync generator is a desirable feature since the video switcher's internal timing is not user adjustable. In the Edit 12 configuration, the secondary sync generator provides flexibility. In the case of RGB chroma key inputs, which are not auto timed, the secondary sync generator could be used to accommodate them. The secondary generator could also be used to allow more timing flexibility in a networked DVE configuration.

Monitor Router

Edit 12 includes a 20 x 10 analog video router. This router's main purpose is to route the character outputs of each VTR to the edit room's monitor wall. The character outputs provide an analog video signal keyed with time code. A control panel for the 20 x 10 is located in the monitor turret at the technical directors position.

Edit 12's use of VTR character outputs will be discontinued once the Television City router project is completed. The router project includes a 100 x 140 longitudinal timecode router. This router, used in conjunction with external time code readers, will provide monitor wall time code readouts as well as burned in time code. The analog monitors presently installed in Edit 12's monitor wall will be replaced with cost effective serial input models. The new monitors will receive serial digital video feeds from the active loop-throughs on the Edit 12 video switcher.

Although Television City presently uses longitudinal time code, it is hoped that either VITC or another form of embedded time code will become standard. Using embedded time code would allow a new type of digital picture monitor to extract the time code and display it. This would make serial digital edit room design even more efficient, eliminating the need for a separate time code or character router.

EQUIPMENT SELECTION

The functionality of the equipment selected for Edit 12 was critical. With most of the equipment being recently released, it was important to find out exactly how the equipment worked. Field trips to various facilities provided the necessary information. The Equipment selected by Television City was judged on the basis of functionality, reliability, availability, and price. Table 1 list the major equipment supplied for the Edit 12 project.

Table 1. Edit 12 Equipment List

1. Video Switching

    Grass Valley Group 3000, D2 serial production switcher
    32 serial inputs, 16 analog inputs,
    2 ME banks, 8 analog aux buses, 4 serial aux buses
    PGM, DSK-key, ME1-2, and SW-PVW analog outputs
    3 Serial Digital PGM outputs

    Datatek 20x10 analog monitor router
    11 Datatek D609 analog video DAs

2. DVE

    Grass Valley Group Kaleidoscope
    2 channels, combiner, kurl, defocus output router
    6 Datatek D609 analog video DAs

3. Video Tape

    4 Sony DVR-20's w/internal mux/demux board BKDV-105,106
    Sony BVH-2000s

4. Editor

    Grass Valley Group VPE 251 w/digital Aux bus software
    Port expansion frame

5. Audio

    Graham Patten DESAM 800, digital audio mixer
    24 serial inputs, 16 analog inputs
    Serial PGM output, Analog PGM output, Analog monitor output

Crown PS-200 audio amplifier
2 JBL 4425 monitors
4 Wohler AMP-1A audio test monitors
2 Datatek D531 analog audio DA's

6. Pulse and Test
    GVG 9520A NTSC sync generator
    GVG 9505A NTSC sync generator
    Tek TSG170D signal generator w/serial output
    2 Tek 1720 Vectorscopes
    2 Tek 1730D Serial Waveform monitors

7. External Multiplexing and Demultiplexing
    5 Sony BKPF-106 Demultiplexers
    1 Sony BKPF-105 Multiplexer

8. Serial Digital Modules
    1 GVG M9132 Reclocking Serial Fanout DA

9. Video Monitors
    1 Sony BVM 1910 with serial input
    2 Sony BVM 1310's with serial input
    6 Sony PVM 1342Q 13"monitors
    6 Sony PVM 8041Q 8" monitors
    2 Sony 5" triple pack monitors

10. Communication
    RTS MCE325 2 channel intercom

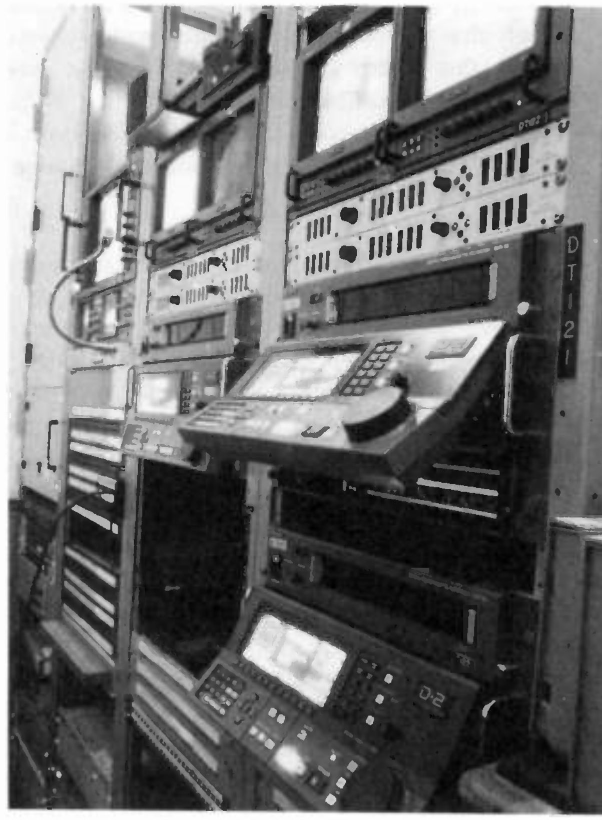
11. Jacks and Coax for Multiplexed A/V
    Belden 8281B Coax
    Kings 7500-1 Video Jacks

12. Jacks and Twinax for AES/EBU serial audio
    Trompeter TWC-124-2 TWINAX
    Trompeter J72 Twinax Jacks
    Trompeter PL75C-214 Concentric BNC

EDIT 12 CONSTRUCTION

The design, construction and integration of Edit 12 was completed in 4 weeks. This was a relatively short time span considering both facility and electronic construction were required. Due to the multiplexed serial design, many hours of documenting, wiring, and integrating were saved.

In the past, TV City edit rooms have required over 40 drawings for proper documentation.



Edit 12 required only 9 drawings. This was accomplished in a variety of ways, all of which are inherent and unavoidable in a serial digital edit room design. Unlike traditional edit rooms, Edit 12 uses no audio or video connection blocks. In a serial digital design, audio and video blocks serve no purpose. If used they may introduce reflections and/or errors into the high speed data stream. Multiplexed digital audio, carried simultaneously with video, also reduces documentation and wiring complexity.

The Edit 12 audio/video block diagram was drawn on AutoCad using a single sheet of D size paper. Although the drawing is intended to be viewed as a complete Edit 12 picture, layering was used to separate the audio, video, and pulse systems. The TV City maintenance department has indicated numerous times that the Edit 12 block diagram, complete with all layers, is very effective and easy to use. At one glance all system connections can be seen. The TV City Engineering Construction group built Edit 12 using only 3 drawings, the A/V block diagram, the serial video jackfield, and the serial audio jackfield. As audio and video become recognized as digital data and not individual systems, the documentation methods used in Edit 12 will become more common.

Edit 12 uses NEC rated, Belden 8281B, for serial video and multiplexed A/V. As mentioned in a previous section, timed cables are not required in a serial digital edit room design. Timing problems in a serial D2 edit room are generally associated with equipment, not cable length. Coax cables in Edit 12 were cut to similar lengths, but not timed, this resulted in a considerable cost savings. To prevent coaxial cable reflections, only one 75 ohm self normalling jack pair is used between a source and it's destination.

Trompeter 120 ohm twinax was used for balanced serial audio. Concentric 120 ohm tip jacks were used at the inputs and outputs of the audio mixer. The twinax cable and Concentric jackfield proved easy to wire and use. The jackfield connects to the twinax cables using concentric BNCs. This type BNC-twinax connection provides full EMI shielding while at the same time allowing the use of balanced audio. Presently there is talk in the industry of standardizing the digital audio connector to 75 ohm BNC's. Although it is the authors opinion that this will work, the methods and materials used in Edit 12 have merit and should be considered.

CONCLUSION

Although serial digital equipment is presently more expensive than parallel or analog equipment, increased hardware cost can be accommodated through simplified system design and construction techniques inherent in a multiplexed serial digital project. The multiplexing performed in Edit 12 is truly transparent to the user and dramatically simplifies the system design. The long term savings in a serial digital project are even more compelling since reduced maintenance cost, and simplified operation are compounded over the life of the project.

It is hoped that the information and ideas presented in this paper have helped the reader to better understand how a multiplexed serial digital edit room works and that it can be built and operated with relatively few problems.

ACKNOWLEDGEMENTS

The author would like to thank George Benkowsky, Director of TV City Engineering, Dave Conant, Director of TV City Technical Operations, Gary Gillett, Manager of TV City Facilities, and Dan Sullivan, Manager of TV City Video Tape Operations for their guidance and many useful discussions. He would also like to thank John Joannou for his detailed assistance during the Edit 12 project.

AM/FM IMPROVEMENT

Sunday, April 18, 1993

Moderator:
James Ary, Great American Communications, Columbus, Ohio

ECONOMIC METHODOLOGY FOR AM SYSTEM TUNEUP
Thomas Gary Osenkowsky
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Brookfield, Connecticut

A NEW MEDIUM-WAVE RADIO TRANSMITTER USING HYBRID MODULATION
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DSP BASED RECEIVER WITH DPLL AND ENHANCED FREQUENCY RESPONSE FOR AM STEREO RADIO
Sangil Park and Dion Messer Funderburk
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A DIGITAL APPROACH TO AN FM EXCITER
Ronald C. Frillman
Harris Allied Broadcast Division
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SELECTION OF FM ANTENNA ELEVATION PATTERNS
Karl D. Lahm, P.E.
Consulting Engineer
Fairfax, Virginia

***SPECIALIZED SIGNALS FOR ADJUSTMENT AND EVALUATION OF BROADCAST AUDIO TRANSMISSION SYSTEMS**
John Bisset
Multiphase Consulting
Falls Church, Virginia
Christopher Downing
International Tapetronics
Bloomington, Illinois

*Paper not available at the time of publication.

ECONOMIC METHODOLOGY FOR AM SYSTEM TUNEUP

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ABSTRACT

Many AM broadcasters find themselves at a disadvantage when it comes to competing with FM in terms of fidelity and coverage. In some cases, improvements can only be made at great expense due to the age of equipment or poor initial design.
In many cases, substantial improvements can be achieved with little cost provided the station's engineer is willing to invest some time and effort in the project. Cited below are actual case histories of three AM directional stations where a considerable gain in operating performance was achieved while total project expense was kept to a minimum.
This presentation is mostly kept in "storytelling" form so that the reader can apply the experiences here to his/her own facility(s).

CASE #1

The first case to be examined is a three tower dogleg array with different day and night patterns. The day operating parameters are listed in Table 1.

Table 1

Twr	F Ratio	Phase	Spacing	Orient	Height
1	.437	-139.5	72	340	126
2	1.000	0.0	17	250	115
3	.553	+145.5	72	160	126

This station came on the air in 1969 using towers 1 and 3 from an existing array that operated on a lower frequency. Tower 2 was added to form the "dogleg" offset array and accommodate higher power (5 kw) on the new frequency. The original 1969 phasor design was based on the calculated base impedances and parameters shown in Table 2.

Table 2

Twr	Base Z	Base I	Base Phase	Power
1	-50+j236	4.45	-142.5°	-991.7
2	50+j134	10.2	-3.0°	5275
3	23+j35	5.63	+142.5°	720

Using moment method computer modeling in conjunction with measured self and mutual impedance data, we show the new values in Table 3.

Table 3

Twr	Base Z	Base I	Base Phase	Power
1	-2016+j1221	.675	+157.5°	-920
2	133+j356	6.44	0.0°	5515
3	44.3+j382	3.02	+138.5°	405

Note the drastic difference between the 1969 predicted values and the 1990 computer generated values. The 1969 phasor design specified two transmission lines each -55° in electrical length between the end two towers back to the #2 center tower where the phasor is located. Considering that the end two towers are 72° apart from tower #2, the lines would have ended up a bit short. Measuring the transmission lines using a signal generator and oscilloscope verified the lines to the -83° for tower #1 and -101.8° for tower #3.

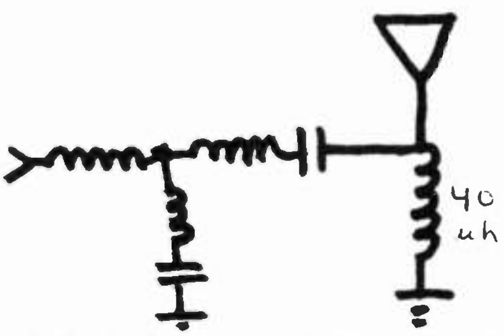
It might seem that the best solution to this array's problems would be a new phasor and ATU's. An inventory of all RF components in the antenna system was taken in order to determine if rearrangements could be made to more produce a more efficient day and night pattern. Knowing the drive point parameters and line shifts allowed the use of electronic circuit analysis computer programs to aid in the redesign of the existing phasor.

In this array, sample loops located 15 feet above each tower base feed equal length lines back to the antenna monitor. Due to the height of the radiators, the placement of the sample loops makes array tuneup difficult. Consider tower the tower #1 current and phase distribution from the base up to 43 feet shown in Table 4.

Table 4

Feet Above Base	Phase	Current
0	+157.5	0.675
21.4	-159.4	2.126
43.0	-154.2	3.120

Note the drastic change in phase as you go from the base up to 43 feet. This is a very unstable area and not a good sampling location. Since relocating the loops would have involved a tower man and the purchase of isolation coils, it was decided to leave the sampling system as-is. The coupling network for tower #1 was redesigned and appears as Figure 1:



A 40uh coil shunts the tower and thus reduces the 2016+j1221 impedance to a more manageable 37.2+j318. A conventional Tee network matches this drive point impedance to 50 ohms. The system phasing was carefully chosen so that both the pattern and impedance bandwidth of the day and night patterns are optimized. As a matter of interest, let us look at the current and phase distribution for tower #1 in the night mode in Table 5:

Table 5

Feet Above Base	Phase	Current
0	+174.9	1.314
21.4	+173.1	2.014
43.0	+172.5	2.412

Notice that the current and phase distribution is a function of drive parameters and that the phase is not constant about the tower as sinusoidal theory implies.

The question of knowing what the correct daytime base operating parameters for tower #1 were given the unpredictable nature of that region was resolved by setting up the phasor and ATU coils using an OIB. The computer generated values were used to set up the phasor and coupling networks. While the accuracy of using a bridge to set up coils has been debated in the past, the method served sufficiently well to limit final touch-up time to one-half hour. Previously, tower #1 ATU presented a load of 0+j5 to the transmission line. Upon initial turn on after the modifications, ATU #1 presented a load of 52+j10. Not bad for initial startup. The monitor points were trimmed as were the remaining transmission line and common point impedance within the one-half hour time period.

The cost of this project was kept to a minimum by careful planning. All transmission lines were measured for electrical length. Base self and mutual impedances were measured. Modern computer design techniques were employed so as to obtain optimum performance given the available parts on hand. Only one new coil and capacitor were purchased. The night pattern was optimized by the same techniques and required no new parts. After two years of operation, the arrays have been stable and virtually trouble free.

CASE #2

An existing station located in Connecticut opted to replace their two guyed towers with self supporting towers so as to keep the entire transmission facility within their own property. Land was previously leased for guy anchors for the old array. When the towers were replaced, power was increased from 500 watts to 2500 watts.

A new phasor and ATU's were purchased from the transmitter manufacturer. The phasor was delivered as factory pretuned. Upon initial turn-on, the phase angle was found to be over 100° in error. When an error this severe is encountered, some investigation is necessary before any adjustments are made.
When the tower crew mounted the ATU's, it was immediately discovered that they were reversed. This situation was rectified and installation of the ground system was completed. Initial investigation revealed that the transmission lines were of the correct length, sample lines were likewise correct and the phasor was checked using a low power oscillator as the signal source and the antenna monitor as a load. The correct phase relationships and current (voltage) ratios were generated at the transmission line output terminals.

The ATU's were checked with an impedance bridge and found to have the expected reactances. Next, the base self impedances were measured. The non-directional mode design specified a base impedance of 40+j45. Measurements indicated a value of 42.5+j55. The ATU output legs were readjusted to compensate for this change, and in conjunction with a minor change in the shunt leg of each ATU, produced a phase angle within 2° of design value.

This actual case is an excellent example of how four hours of investigative time using instruments and calculators solved what turned out to be a relatively simple problem. The case also proves that an array that would appear to be far out of adjustment may in fact only require a minor modification. Had the phasor and ATU's been randomly adjusted to produce the correct pattern, impaired bandwidth and poor efficiency would have resulted.

CASE #3

The last case to be discussed is a four tower DA-2 array consisting of 190° radiators. Towers #1 and #2 have top mounted FM antennas and tower #2 also has a RPU antenna mounted at mid-point. Sample loops are located 200 feet above each tower base with the isolation coils located inside each ATU. This array had numerous problems, most notably antenna monitor parameters and monitor point field intensities out of tolerance.

The "normal" antenna monitor readings that were posted near the instrument dated back to 1977, the time of the latest complete proof of performance for this array. A partial proof was performed on the day and night patterns in 1981 with new array parameters and monitor point limits assigned by the FCC. This was a serious problem because the station operators, as well as the Chief Operator, were not aware that they had been referencing the array performance to outdated parameters. Furthermore, the phasor had been "tweaked" back to these old parameters as the seasons changed.

Initial day pattern values read on the antenna monitor varied considerably from the 1981 licensed values. Table 6 shows the indicated parameters while Table 7 shows the 1981 daytime licensed values.

Table 6

Twr	Ratio	Phase Angle
1	.375	-148°
2	1.000	0.0°
3	.200	+142°
4	.175	+169.5°

Table 7

Twr	Ratio	Phase Angle
1	.860	-145°
2	1.000	0.0°
3	.600	+144°
4	.175	+175°

The problem with tower #3 was believed to be in the sampling system since the night ratio reading was far below licensed value. The ATU at each tower base consists of three coils and two or three capacitors in common with day and night pattern. The coil taps are selected via a contactor for each mode of operation.

Examination of the sample line connectors in tower #3's isolation coil revealed burning at the outer sheath connection inside the "N" connector. The outer sheath was never bent back on the connector body to form a solid ground electrical connection. Trimming the sample line and reattaching the connector restored normal sample indications to tower #3. Where isolation coils also serve as static drain chokes, it is important to keep in mind that both RF and static charges flow in the isolation coil outer conductor. The connector in question failed due to excessive arcing and burning over a period of time.

The problem with tower #1 was traced to a bad capacitor in the day phasor. Due to severe mismatches in the transmission lines, the failed capacitor was subjected to excessive current over an extended period of time. One cannot ignore the component ratings in a phasing and coupling system. Where doubt exists, it is wise to measure the operating *impedance* that a component is subjected to, not just the resistance. An ammeter can be used to measure the current flow thus determining the power in the circuit. The voltage can be solved for using Ohm's Law for AC circuits.

While examining the array for other problems, it was noted that tower #3 again showed signs of instability. Lightly banging the ATU caused a phase change of 2° and a ratio change of 1.5%. The problem was traced to the ATU's inner door hinges making intermittent electrical contact. The instability was solved by attaching copper braid from the ATU door to the inside cabinet. This was done at each ATU as a precautionary measure.

Since there were numerous problems throughout the system, I used a standard plan of attack to bring the array within specifications. The plan is as follows:

1. Measure each transmission line and sample line for electrical length using an RF oscillator connected to the phasor end with the ATU end open circuited. The oscillator is tuned to 300 Khz or so and slowly swept upward until a null is observed at the phasor end. This frequency is recorded. This procedure is repeated until the second highest null frequency is found.

In the case of the sample lines, the far end is shorted due to the fact that the sample loop is a DC short. The electrical length is found from Equation 1 for the open circuited end or Equation 2 for the short circuited end.

2. Measure and record each tower's self and short circuited impedance. This procedure calls for opening the RF feed to each tower and measuring each tower's self impedance. The remaining towers are each shorted to ground, one at a time and each impedance value recorded. Equation 3 is used to determine the value of mutual impedance. Care must be used in Equation 3 since there are two possible correct answers. One must consider the element spacing to account for the proper quadrant sign of the impedance values.

3. Perform a moment method computer analysis of the array using the measured self and mutual impedances as a guideline for building the array model.

Once the model is constructed, a phasing and coupling system can be designed using as many, if not all, of the existing inventory of parts. Any additional parts can be weighed on an improvement versus cost basis.

4. Once the array is tuned up and trimmed, a partial or complete proof of performance should be executed and filed with the FCC.

In Case #3 it was found that the computer model did not produce the expected results in the field. While it was found that towers #3 and #4 had base impedances of 75-J205, towers #1 measured 50-j152 and tower #2 measured 44-j129. Towers #1 and #2 have FM antennas and isocouplers across the bases. These were accounted for in the computer model. During further investigation, I noted that the guy wires appeared to be excessively long given the station's frequency. In fact, one of the lower guys had a long rubber sleeve placed on it. The station often threw picnics in the antenna field and guests were getting RF burns from the guy wires.

Normally, one-eigth wavelength is the maximum length used in guy wires in the AM band. In this case, the guys were 35% longer than normal with only one insulator used between lower spans. This produces a reradiation filed within the array and causes the cylindrical model used in the moment method computer analysis to be unrealistic.

A new moment method tower model was used for each radiator and the ATU's were readjusted to compensate for the difference.

The feeder system was adjusted to the computer generated operating parameters. The parameters shown on the antenna monitor reflected the differences in sample line length determined in step #1. Even in "equal length" sample systems, the lengths should be verified first. If sampling transformers are used or if sample loops are located near the tower base, it is a wise precaution to open the far end of each sample line and "meg" it to see if there is any leakage from the center conductor to the outer sheath. Sampling transformers can be checked for resonant frequency using an oscilloscope and RF oscillator. Sampling transformer problems are usually isolated to a shorted varistor which is placed across the "N" connector.

When measuring the monitor points, I always carefully follow the licensed directions to each point, noting any discrepancies along the way. Since this station's original proof, some of the road names have changed and the oak trees used as a reference for one of the points are no longer in existence. This kind of discrepancy can easily lead to an FCC citation. When filing the partial proof, new point directions, descriptions and /or photographs should be included to make certain that each point is well depicted.

Proof measurements can be especially difficult when accurate point descriptions no longer exist. The older the station's original proof, the more difficult the task. Careful examination of the topographic maps used in the original proof can be very helpful. Often times street names have changed, pastures now have condominium complexes built on them and shopping malls have overrun once wooded areas obliterating monitor points. In some of these cases, no other choice exists except to do a new complete proof of performance.

At the time of this writing, the proof phase of Case #3 has just begun. It is anticipated that no major problems will be encountered. Proof analysis can be a challenging task, especially considering the rapid pace of environmental changes that seem to occur on a daily basis. In two other cases, it was found that the radials laid out in the original proof were 2° and 3.5° off from their True North heading. In the latter instance, this was discovered while plotting the radials onto new topographic maps since the originals were destroyed in a flood. The original plots were taken from the reference proof on file with the FCC.

CONCLUSION

It is often said that experience is the best teacher. While it is not possible to relate a career's worth of experience in a short writing, it is hoped that some of the material presented in this paper can be applied to AM arrays that are maintained by the readers.

Equation 1

K = 2 / [(F hi / F low) - 1]

Length° = (F carrier / F low) * K * 90

Equation 2

K = 2 / (F low / F high)

Length° = (F carrier / F high) * K * 90

Equation 3

$$Zp,q = Zq,p = \sqrt{(Zpp \times Zqq) - \left(Z\tfrac{p}{q} \times Zpp\right)}$$

A NEW MEDIUM-WAVE RADIO TRANSMITTER USING HYBRID MODULATION

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ABSTRACT

Described is a new AM radio transmitter with an output of 1kW using hybrid modulation which can produce AM wave directly from digitalized audio signal by combining the outputs of switching type RF power modules and one analogue type power module. Use of this system results in an transmitter with outstanding high over-all efficiency (86% or more) and excellent audio performance.

INTRODUCTION

This paper presents a new MF transmitter using " hybrid modulation " which can produce AM wave directly from digitalized audio signals by combining the output of switching type RF power modules and one analogue type RF power module.

Although AM transmitters using PWM have been mainly used in broadcasting service, their drain efficiency was approximately 60%[(1)~(3)] at most.

Use of this system results in an transmitter with outstanding high over-all efficiency (86% or more) as full digital transmitters[(4),(5)] and cost-performance.

FEATURES

The new transmitter has following features.

(1) High reliability

This AM transmitter is constructed by employing a new type of modulation (hybrid modulation) which can produce good quality AM with a simple configuration.

(2) High efficiency

Over-all efficiency of 86 % or more is obtained, which can reduce power costs significantly.

(3) High sound quality

The transmitter displays signal-to-noise ratio of 72dB or better and total harmonic distortion of 0.4% at 80 % modulation.

THEORY OF OPERATION

Fig. 1 shows the block diagram of the transmitter. The transmitter consists of sixteen switching power modules, one analogue type power module, 12 bits

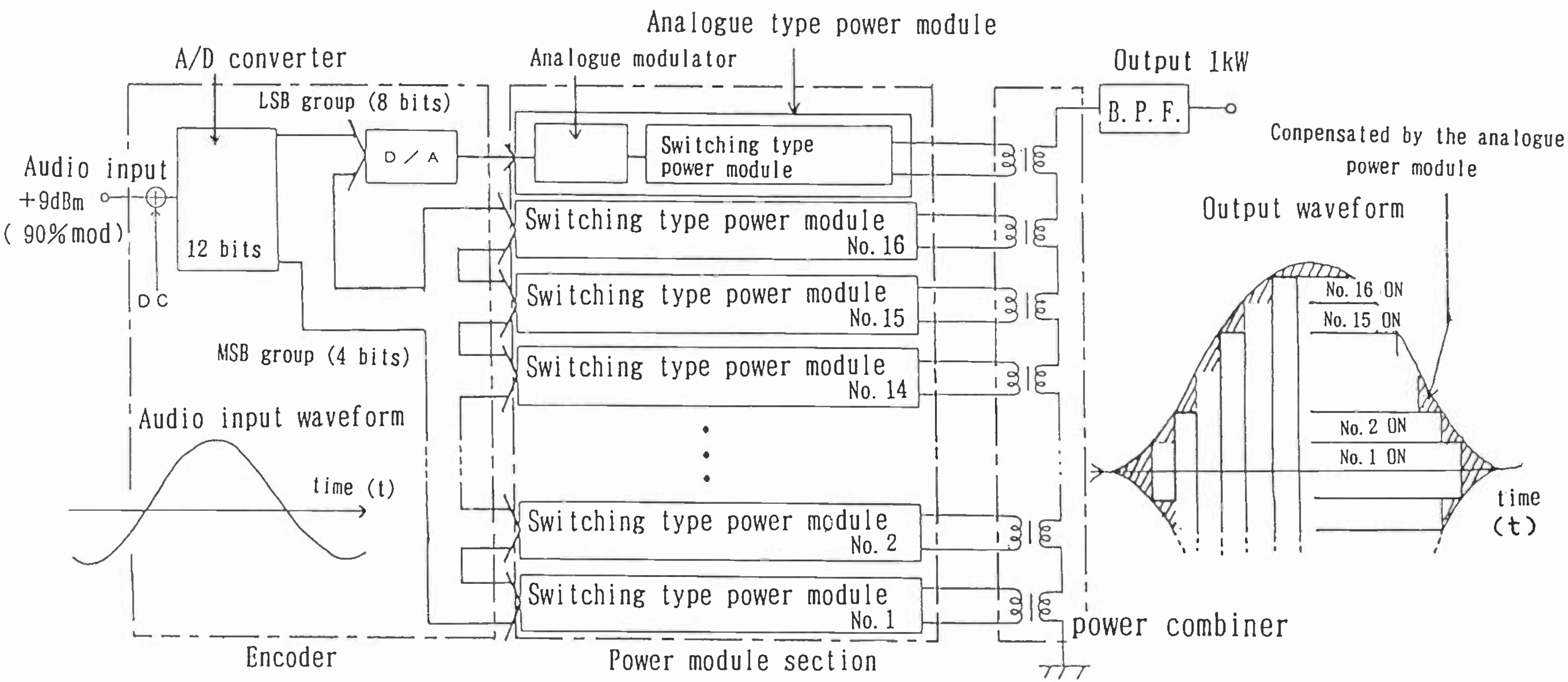


Fig. 1 Block Diagram of the Transmitter

and D/A converter which reconverts digital information (LSB group 8 bits) back to the analogue signal to modulate the analogue power module.
Digitalized audio signals are divided into MSB group bits information, and one analogue type power modulated according to the LSB group bits.
AM wave is obtained by combining the output of the switching power modules and the analogue type power module.
Use of the analogue type power module for LSB group 8 bits results in an small number of power modules.
The external view of the transmitter is shown in Fig. 2. The transmitter is 570(W)×2060(H) ×450(D) mm^3 in size.
Fig. 3 shows the basic circuit configuration of the RF power module and power combiner with an output impedance of 6.4Ω.
The RF power module of dual SEPP configuration, having a peak output power of 280W and output impedance of 80Ω, features a drain efficiency of 95% at a drain-source voltage of 120V.
The performance of the MOSFET used in the RF power modules is shown in Table 1.
A MOS-FET having the ON-resistance as low as possible, is employed in order to obtain a high drain efficiency.
External view of the RF power module and power combiner are shown in Fig. 4 and Fig. 5 respectively.

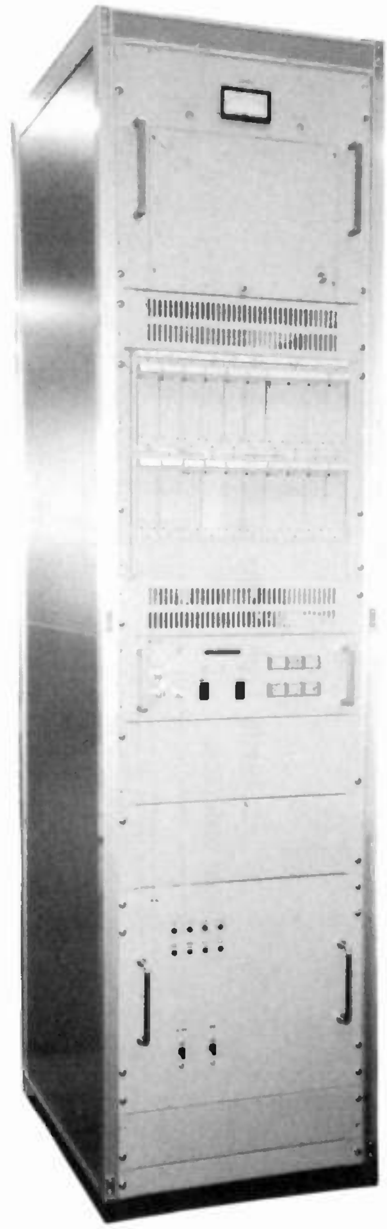


Fig. 2 External view of the transmitter

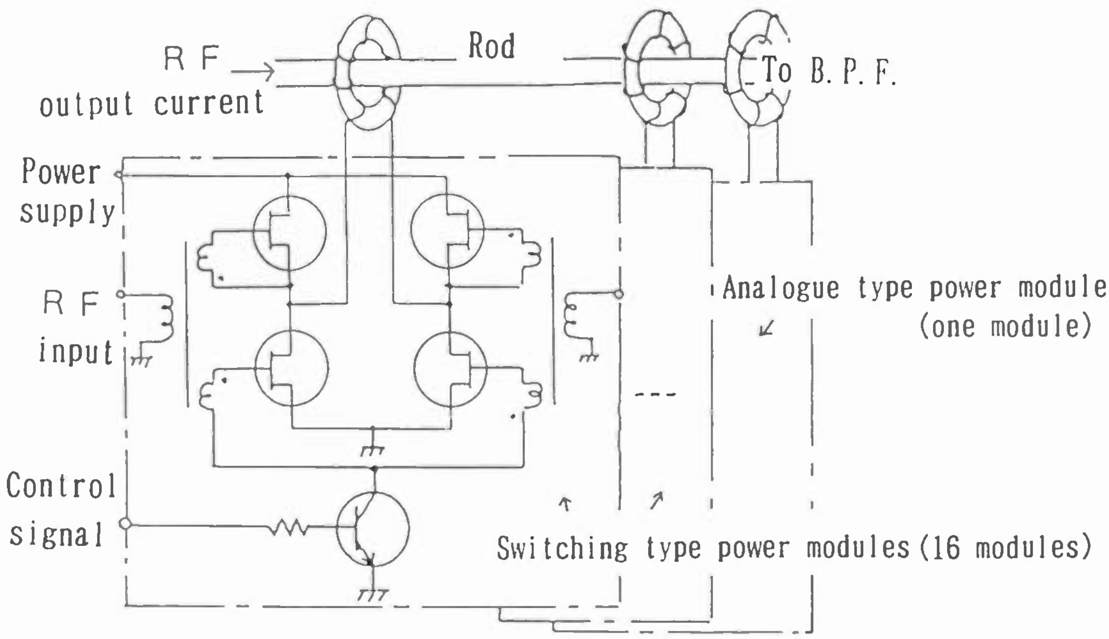


Fig. 3 Basic circuit configuration of the RF power module and the power combiner

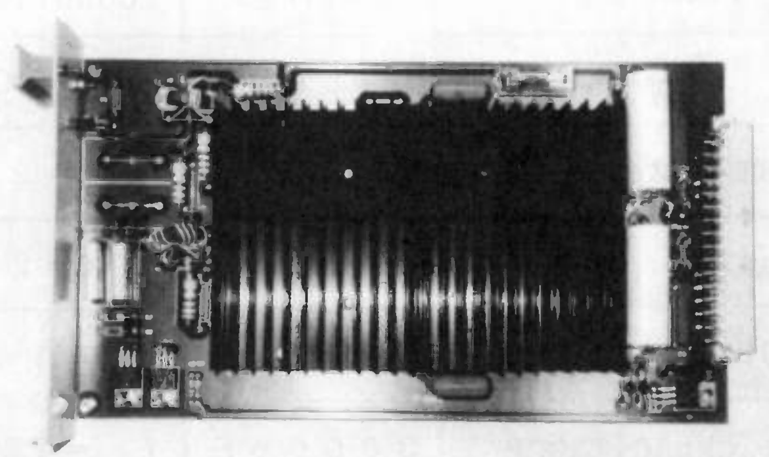


Fig. 4 External view of the switching type power module

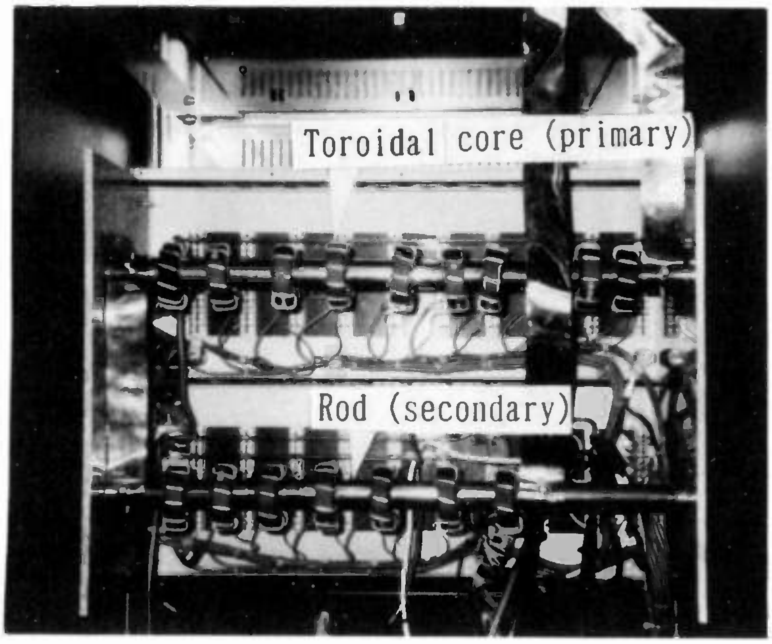


Fig. 5 External view of the power combiner

Table 1 Performance of the MOS-FET

Items	Ratings	Conditions
Drain-Source Breakdown voltage	450V	VGS=0
Maximum drain current	20A	TC=25° C
Allowable loss	150W	TC=25° C
Output capacitance	900pF	VDS=10V VGS=0 V f =1MHz
Input capacitance	3000pF	
ON resistance	0.25Ω	ID=10A

PERFORMANCE

Table 2 shows the ratings and performance of the transmitter.

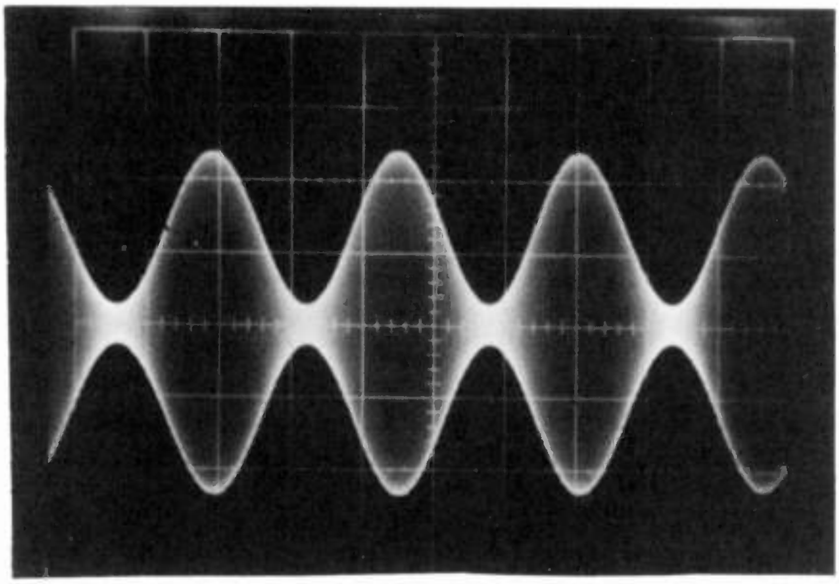
The transmitter displayed an overall efficiency of 86% or more at a frequency of 1 MHz, low distortion of 0.4 % or less and a signal-to-noise ratio of 72dB.

Output signal waveform of the transmitter for generating an output of 1 kHz is shown in Fig. 6.

Fig. 7 shows the output signal waveform of the transmitter without the

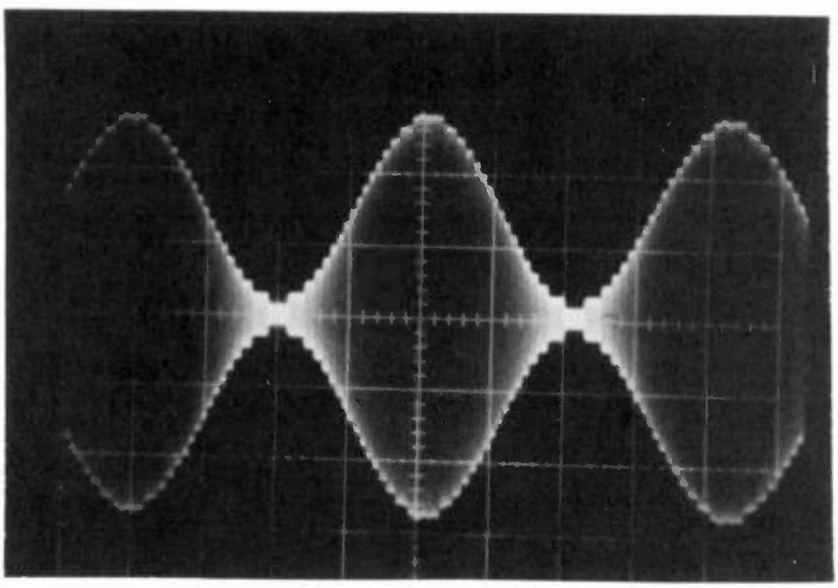
Table 2 Ratings and performance of the transmitter

(Ratings)	
Output power	1kW
Frequency range	531kHz ~1602kHz
Output impedance	50 Ω
Audio input	+9dBm
Audio frequency range	50 ~7500Hz
Cooling system	Forced-air cooling
Power supply	AC 100V/200V, single phase
(Performance)	
Overall efficiency	85 % or better
Audio frequency response	±1dB or less, 50~7500Hz
Carrier shift	3.5% or less at 1kHz, 80 % modulation
Distortion	2.5% or less at 20~80% modulation, 50~7500Hz
Signal-to-noise ratio	60dB or better at 1kHz, 80% modulation
Stability against the change of the load condition	Continuous operation at VSWR=1.5 and 100 % modulation



(V : 250V/div. , H : 0. 38msec/div.)

Fig. 6 Output waveform of the transmitter



(V : 250V/div. , H : 0. 25msec/div.)

Fig. 7 Output waveform of the transmitter without the analogue type power module

analogue type power module.
Its distortion was 4.7% because of the saw-tooth waveform.

CONCLUSION

The new 1kW AM transmitter having the superior performance is constructed. Since this transmitter offers such high efficiency and reliability, this technology will spread from 1kW to 500kW or more transmitter in the near future.

ACKNOWLEDGMENT

The authors wish to thank Japan Radio Corporation for constructing this new transmitter.

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DSP BASED RECEIVER WITH DPLL AND ENHANCED FREQUENCY RESPONSE FOR AM STEREO RADIO

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Abstract - **This paper presents a brief background of AM stereo technology followed by a discussion of a digital implementation of an AM stereo receiver. The discussion includes the derivation and implementation of digital demodulation, filtering, and stereo decoding. A digital phase lock loop (DPLL) is described to assist in the digital demodulation process. A filter which compensates for the poor frequency response of the analog front end is presented as well. This implementation results in a superior AM stereo signal to that of the stereo signal received by a typical analog receiver.**

I. INTRODUCTION

Digital audio systems with DSP (digital signal processing) are the latest technology in automotive as well as home audio systems. These systems use DSP to create concert hall effects, reverberation and noise cancellation as well as CD quality sound. The DSP's use can be extended to that of digitally receiving radio signals as well, which reduces analog circuitry in audio systems while providing a superior quality sound. Although Digital Audio Broadcast (DAB) is not yet available, this trend to digital radio receivers for the standard analog audio signal is a step in the direction of completely digital radio systems. The techniques used in digitally recovering the analog signal will certainly be useful in designing a receiver for the DAB signal even though the modulation technique is much different.

The authors derived and implemented an algorithm to perform AM radio reception with and without the C-QUAM® (Compatible Quadrature Amplitude Modulation) stereo signal. This paper outlines this derivation, analysis and implementation of the digital radio using digital signal processing techniques.

† C-QUAM® is the registered trade mark of Motorola, Inc.

II. Background

The FCC approved a stereo system for FM broadcast at the peak of AM radio popularity. When this occurred in 1961, AM radio had the biggest share of the radio market [1]. The stereo sound that FM provided quickly eroded the AM radio market and efforts were made to create a stereo broadcast system for AM. Many techniques were developed to perform AM stereo transmission to recover a larger share of the radio audience lost to FM.

The FCC never specified one particular AM stereo technique as a standard as they did with FM, but did specify that every AM stereo broadcast technique must allow installed monaural AM receivers to receive an undistorted signal when tuned to a stereo channel. As much as this complicated the design of AM stereo techniques, the FCC tested and approved several standards including the C-QUAM® technique in 1981 [2]. The marketplace has made the C-QUAM® technique the *defacto* standard for AM stereo transmission with approximately 574 stations throughout the United States and 206 stations internationally for a total of 780 C-QUAM® stations world wide using C-QUAM® encoding [3].

The most obvious way of providing a stereo signal with AM transmission would be to modulate a carrier with the $L + R$ signal and a 90 degrees shifted carrier with the $L - R$ signal. Due to the FCC imposed compatibility requirements, using this typical QAM system for transmitting $L + R$ and $L - R$ as two different signals as shown in Figure 1, is not possible. An envelope detector would not be able to detect this signal if a large modulation index existed because a stereo signal in which L only or R only was transmitted. Thus, it would not be compatible with existing receivers which use envelope detectors.

To create a stereo signal, the C-QUAM® technique transmits L (left channel) and R (right channel) infor-

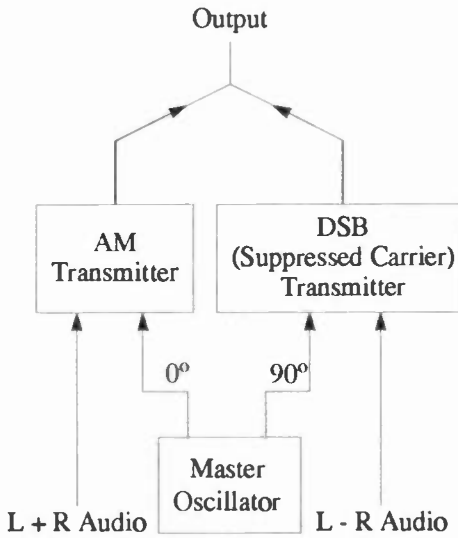

Figure 1. Quadrature AM Stereo Transmitter

mation by encoding $L + R$ and $L - R$ channel information into the phase of the transmitter carrier. This "modified" carrier is then amplitude modulated by the $L + R$ signal [4], as would be done in a typical monaural AM transmitter with an unmodified carrier. Figure 2 shows the carrier phase modification of a C-QUAM® transmitter in block diagram form. The output of the I and Q modulators, respectively are:

$$I = [L(t) + R(t)] \cos(\omega_c t) \tag{1}$$

and

$$Q = [L(t) - R(t)] \sin(\omega_c t) \tag{2}$$

Summing these signals with additive carrier signal result in

$$I + Q + \cos(\omega_c t) = [L(t) + R(t)] \cos(\omega_c t) + [L(t) - R(t)] \sin(\omega_c t) + \cos(\omega_c t) \tag{3}$$

which can be expressed as

$$B(t) \cos[\omega_c t + \gamma(t)] \tag{4}$$

where

$$B(t) = \sqrt{[1 + L(t) + R(t)]^2 + [L(t) - R(t)]^2} \tag{5}$$

and

$$\gamma(t) = \tan^{-1}\left(\frac{L(t) - R(t)}{1 + L(t) + R(t)}\right). \tag{6}$$

Then the signal is hard limited by the limiter shown in Figure 2 which yields unchanged phase but a constant amplitude signal as

$$A \cos[\omega_c t + \gamma(t)] \tag{7}$$

where A denotes the pre-defined constant. This signal essentially becomes the carrier which is modulated by the $L(t) + R(t)$ signal. The resulting signal from the transmitter is:

$$T_x(t) = A[1 + L(t) + R(t)] \cos[\omega_c t + \gamma(t)]. \tag{8}$$

It is possible to receive this on a non-stereo system with an envelope detector with little or no distortion. An envelope detector ignores the phase information contained in the signal [5], so that if the phase component is removed for the signal given above, the result will be

$$A[1 + L(t) + R(t)] \cos(\omega_c t), \tag{9}$$

which is the standard signal which would be seen by the receiver if the transmitter was monaural.

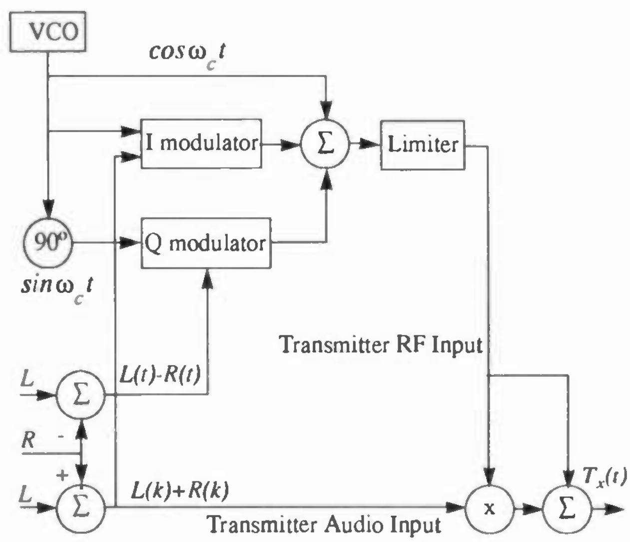

Figure 2. C-QUAM® AM Stereo Transmitter

A block diagram of the analog receiver is shown in Figure 3. The output of the envelope detector is simply the $L + R + C$(carrier) signal normally expected from a monaural system. Using this output and comparing it to the QAM output, the input to the QAM portion of the block diagram shown below can be gain modulated to remove the $\cos\gamma$ term (the input signal is divided by the $\cos\gamma$ term), with the resulting output of the I and Q demodulators being the required signals [3]:

$$[1 + L(t) + R(t)] \tag{10}$$

and

$$[L(t) - R(t)]. \tag{11}$$

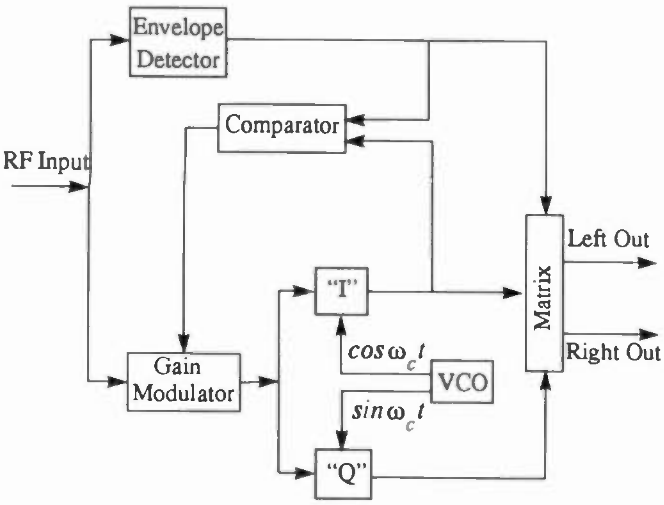

Figure 3. Analog C-QUAM® AM Stereo Receiver

The output of the QAM detector without gain modulation contains the $L + R$ and the $L - R$ encoded signals in the form given below:

$$I(t) = A[1 + L(t) + R(t)]\cos\gamma(t) \qquad (12)$$

$$\begin{aligned} Q(t) &= A[1 + L(t) + R(t)]\sin\gamma(t) \\ &= A[L(t) - R(t)]\cos\gamma(t) \end{aligned} \qquad (13)$$

The two resulting signals (with gain modulation) are added together to produce the left only signal and subtracted to produce the right only signal.

III. Implementation Considerations

Figure 4 shows a detailed diagram of the DSP based system. The input to the system comes from the AM receiver front end and tuning circuitry and has been downconverted to the 450 KHz IF frequency. The sampling rate at the IF frequency would have to be at least 900 KHz to meet the Nyquist criteria [6]. The AM signal is confined to a 10 Khz channel, however, so if the signal is further downconverted, a smaller sampling rate could be used to reconstruct the signal. As shown in Figure 5, the IF signal is downconverted to approximately 25 KHz and a sampling rate of 4 times that is employed in the sigma delta A/D convertor [7]. The analog multiplier shown in Figure 5 creates all ranges of harmonic frequency contents above the fundamental carrier frequency, However, Sigma-Delta A/D converter technology is based on oversampling and decimation processing technology, it eliminates the need to use an anti-aliasing filter [8.9].

The resulting digital signal now becomes the input to a general purpose digital signal processor. All functions from this point on are firmware solutions. The first function necessary is to separate the signal to retrieve the I and Q channels by demodulation to baseband. The digital input samples are multiplied by the current sine and cosine values of the numerically controlled oscillator (NCO). This operation is performed at the input sample rate of approximately 100 KHz. This is at a sample rate that is over 4 times the needed rate for reconstruction of the 10 Khz signal, therefore,

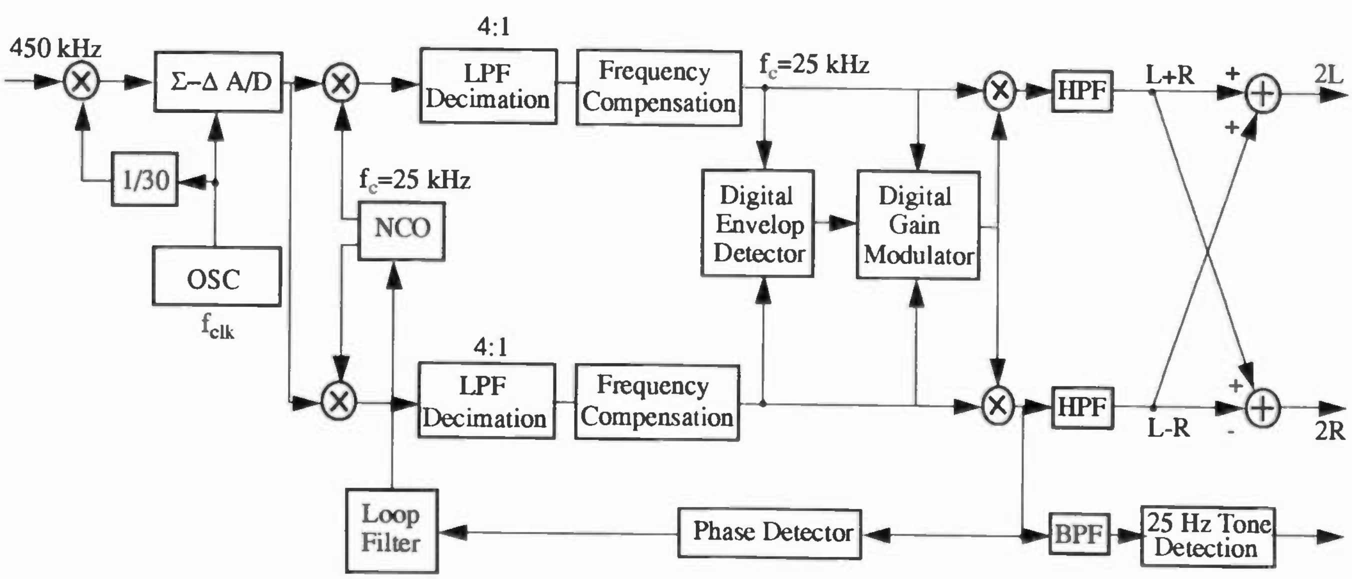

Figure 4. Detailed System Diagram for the Digital AM Stereo Receiver

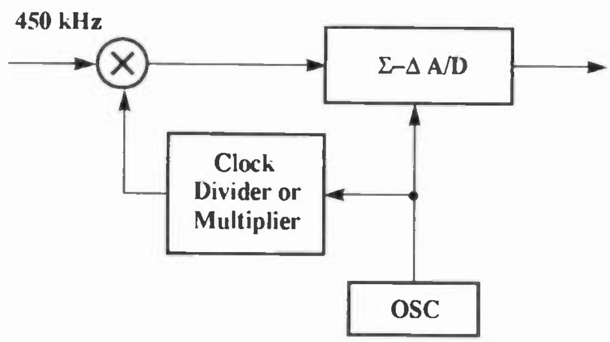

Figure 5. Figure 5 Down-Conversion Scheme

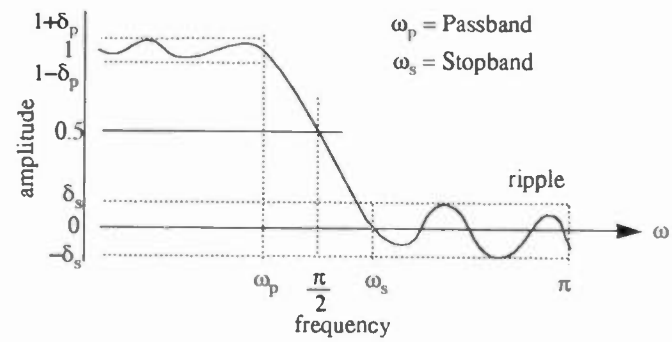

Figure 6. Design criteria for each stage half-band filter

the results of the last operation are low pass filtered and decimated to a sampling rate of approximately 25 KHz.

IV. Halfband FIR Filter Design

Since the phase response is very important to decode the stereo signal and for the DPLL, a linear phase FIR filter is used which utilizes 4:1 decimation and LPF as shown in Figure 5. Consider a special FIR filter called the *half-band* filter, which gives the required filter response and is computationally much simpler than conventional FIR filters. The half-band filter frequency response is symmetric about the half the Nyquist frequency. This constraint results in half of the filter coefficients being exactly zero. Clearly if this symmetry can be tolerated in a design considerable computational complexity can be avoided.

Let $H_h(z)$ be a linear-phase FIR filter transfer function with an odd number of coefficients, N. Since the linear-phase FIR filter has symmetric coefficient values, the frequency response $H_h(e^{j\omega})$ is real-valued function as

$$H_h(e^{j\omega}) = \sum_{n=0}^{M} b_n cos(n\omega) \tag{14}$$

where $M = (N-1)/2$. Assuming the cut-off frequency of the half-band filter is $\omega_c = \pi/2$, we have

$$\omega_p + \omega_s = \pi \tag{15}$$

where ω_p and ω_s denote the passband and stopband edges, respectively. Moreover, assume that the passband ripple and the stopband ripple are the same (*i.e.*, $\delta_p = \delta_s = \delta$). Thus, the response exhibits symmetry around $\pi/2$. Figure 6 shows a plot of such a symmetric halfband FIR filter response. In view of the symmetry, it is straightforward to show that

$$H_h[e^{j\omega}] = 1 - H_h[e^{j(\pi-\omega)}] \tag{16}$$

Also, in terms of H(z),

$$H_h(z) + H_h(-z) = 1. \tag{17}$$

Substituting (14) in (17), it can be shown that the coefficients b_n have the following constraints:

$$b_{2n} = 0, n \neq M$$

$$b_M = h(M) = 0.5 \tag{18}$$

As a result, the impulse response sequence has every odd-number sample equal to zero for M odd (except the coefficient $h(M) = b_M = 0.5$). Since the halfband filter is based on a symmetrical FIR design, the number of multiplications in implementing such a filter is one-fourth of that needed for arbitrary FIR filter designs.

The halfband filter coefficients can be obtained via the Fourier series method using a Kaiser window [10]. By choosing proper parameters in the Kaiser window, more than -120 dB of stopband attenuation with specified transition bandwidth can be realized. Perhaps the only real drawback of the filter design is the requirement that $\delta_p = \delta_s$ shown in Figure 6. For most practical systems we have $\delta_s << \delta_p$. However, since the design curves are relatively insensitive to δ, the computational price paid for designing a filter with δ_p much less than required is generally small, and almost always much less than the 2:1 speedup achieved by these filters.

The halfband symmetric filter has a natural application in decimators with sampling rate changes of 2. However, since a halfband filter can only achieve a 2:1 decimation, a series of such filters should be cascaded to perform a higher decimation filter process. To obtain 4:1 decimation LPF with linear-phase response, a cascaded structure of two half-band filter has been implemented. Figure 7 shows the block diagram of a 2-stage decimation structure.

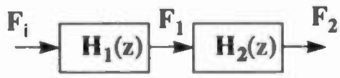

Figure 7. Two-stage cascaded half-band filter structure

F_i in Figure 7 denotes the sampling rate of the input signal to the decimation filter. Note that the decimation of 4:1 can be achieved by two 2:1 decimators in cascade. For this implementation, 11-tap FIR filter is used for the first stage filter, while the other with 31 taps. This is efficient in reducing the number of operations performed per input since the signal is down-sampled to approximately 50 Khz in the first filter using only ll taps. The second filter, which takes 31 taps, is performed at the lower sampling rate.

For decimation, this cascaded half-band filter structure has the following advantages: significantly reduced number of computations; reduced memory requirement; simplified filter design problem; and reduced finite-word-length effects [11].

Let's examine the lowpass filter frequency regions for the individual stages. For the kth stage decimator, when $k = 1, 2$, the lowpass filter's passband, transition band and stopband regions are

$$
\begin{aligned}
&0 \le f \le F_k && : k\text{th stage passband} \\
&F_p \le f \le F_k - F_p && : k\text{th stage transition band} \\
&F_k - F_p \le f \le F_k && ; k\text{th stage stopband}
\end{aligned} \qquad (19)
$$

where F_p and F_k are the passband frequency and the sampling frequency of the kth stage output, respectively. Signal energy in the transition band will alias back upon itself (after decimation by 2:1) only from F_p up to $F_{k-1}/2$; hence the baseband, $0 \le f \le F_p$, is protected against aliasing. The output of this decimation (D) filtering is

$$I_D(k) = A[1 + L(k) + R(k)] \cos(\gamma(k)) \qquad (20)$$

$$Q_D(k) = A[1 + L(k) + R(k)] \sin(\gamma(k)) \qquad (21)$$

where

$$\gamma(k) = \tan^{-1}\left[\frac{L(k) - R(k)}{1 + L(k) + R(k)}\right] \qquad (22)$$

which is the discrete form of the continuous time equations (1), (2) and (6).

V. Compensation Filter Design

Following the decimation filters are the filters which compensate the signal for the poor frequency response of the typical AM receiver front-end. A typical AM receiver front-end is analog and cuts the bandwidth to about 3.5 kHz as shown in Figure 8. This performance can be improved by applying a linear phase frequency compensation filter to the signal at the output of the 4:1 decimation filter stage.

Consider a filter which can compensate the frequency distortion by the front-end analog band-limiting filter. The objective is to design a compensation FIR filter whose transfer function $H_c(\omega)$ is such that its magnitude response approximates the inverse transfer function of the front end and has the linear phase property. Since the resulting FIR coefficients are real numbers, $H_c(\omega)$ is an even function in the interval $-\infty < \omega < \infty$. It is convenient to define the normalized frequency $0 \le \nu \le 1$ where $\nu = f/f_N$, so that the normalized Nyquist frequency $f_N = 1$. Using the Fourier series representation the transfer function can be expressed as [12]

$$H_c(\nu) = \sum_{n=-\infty}^{\infty} h_c(n) e^{jn\pi\nu} \qquad (23)$$

where the Fourier series coefficients $h_c(n)$ is obtained by

$$h_c(k) = \frac{1}{2}\int_{-1}^{1} H_c(\nu) e^{-jn\pi\nu}. \qquad (24)$$

Since $H_c(\nu)$ is an even function in the interval of $-1 < \nu < 1$, (24) can be modified by

$$h_c(n) = \int_0^1 H_c(\nu) e^{-jn\pi\nu} \cos(n\pi\nu)\, d\nu \qquad n > 0 \quad (25)$$

and $h_c(-n) = h_c(n)$. Since the transfer function $H_c(\nu)$ is an unknown arbitrary function, it is extremely hard to find a closed form solution of the integration in (25). A numerical integration technique is used to find a set of coefficients for any shape of transfer function [13]. It is important to divide the given interval into N smaller intervals (Δ), so that the discontinuity between subdivisions becomes minimum. The numerical approximation of (25) can be expressed as

$$h_c(n) \cong \sum_{i=1}^{N} \int_{v_{i-1}}^{v_i} H_c(v)\, e^{-jn\pi v} \cos(n\pi v)\, dv. \qquad (26)$$

Hence, approximating (25) by (26) amounts to finding the solution for the subinterval (v_{i-1}, v_i). There are numerous methods to find the solution [13], however, it has been studied that the Simpson's method gave us the optimum estimation on this implementation.

For convenience, let's define

$$g_i = \int_{v_{i-1}}^{v_i} H_c(v)\, e^{-jn\pi v} \cos(n\pi v)\, dv. \qquad (27)$$

Applying (27) into the Simpson's rule, the FIR filter coefficients $h_c(n)$ defined (23) can be obtained as [13]

$$h_c(n) = \frac{\Delta}{6}\left[g_i + g_N + 2\sum_{i=1}^{N-1} g_i + 4\sum_{i=1}^{N-1} g_{i-1/2} \right]. \qquad (28)$$

However, it is not practical to have an infinite number of coefficients as shown in (23). So the series summation in (23) can be truncated at a predetermined finite number as

$$\bar{H}_c(v) = \sum_{n=-L}^{L} h_c(n)\, e^{jn\pi v} \qquad (29)$$

The truncation in (29) can create the leakage phenomena which can be smoothed by a window function [10]. In this implementation the Blackman-Harris window has been used to optimize for maximum sidelobe attenuation. Using the window function $w(n)$, equation (29) can be rewritten as

$$\bar{h}_c(n) = w(n)\, h_c(n) \qquad for \qquad -L < n < L \qquad (30)$$

where the Blackman-Harris window function is defined as [18]

$$w(n) = 0.358 + 0.488\cos\left[\frac{\pi n}{N}\right] + 0.1412\cos\left[\frac{2\pi n}{N}\right] + 0.011\cos\left[\frac{3\pi n}{N}\right] \qquad (31)$$

Since the frequency response of the compensation filter requires only monotonic changes with respect to frequency, a simple (11-tap) symmetric FIR filter can compensate the frequency drooping to make a flat spectral response as shown in Figure 8. This effectively enhances the frequency response of the signal before further processing, and thus improves the quality of the final audio output. The typical frequency response, the compensation filter response and the combined filter response are shown in Figure 8.

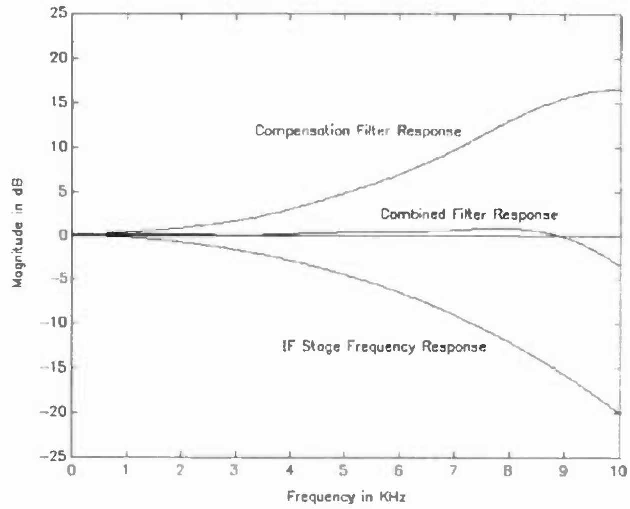

Figure 8. Frequency responses of the IF stage and compensated filter

VI. Detection

Following the frequency compensation (FC) filtering, $I_{FC}(k)$ and $Q_{FC}(k)$ are the inputs to the digital equivalent of envelope detection in the first process of recovering the left and right signals. If the effects of phase error are ignored, then

$$I_{FC} \cong A[1 + L(k) + R(k)]\cos[\gamma(k)] \qquad (32)$$

and

$$Q_{FC} \cong A[1 + L(k) + R(k)]\sin[\gamma(k)] \qquad (33)$$

and $H(k)$ is the output of the digital envelope detection:

$$H(k) = \sqrt{I_{FC}^2(k) + Q_{FC}^2(k)}$$
$$= A(1 + L(k) + R(k))\sqrt{\sin\gamma^2 + \cos\gamma^2}$$
$$= A(1 + L(k) + R(k)) \qquad (34)$$

$H(k)$ can then be used to estimate $\frac{1}{\cos(\gamma(k))}$ where $C(k)$ is this estimate.

An error signal can them be formed since $H(k)$ is known and since the estimate $C(k)$ multiplied by $I_{FC}(k)$ should equal $H(k)$, if the estimate has converged.

$$e(k) = I_{FC}(k)C(k-1) - H(k) \quad (35)$$

The error signal $e(k)$ can then be used to update the estimate by

$$C(k) = \beta C(k-1) + (1-\beta)e(k). \quad (36)$$

β is selected based on the signal characteristics and it controls the rate of convergence of the estimate $C(k)$. A value of 0.99 is typical, and is used in this application.

At the time point, k, it is assumed that the iterative equation of (36) has converged and thus $e(k) = 0$ or, from (35):

$$C(k) = \frac{1}{cos(\gamma(k))}. \quad (37)$$

Thus, I and Q signals can be modified by multiplying by this value of $C(k)$ which results in:

$$I_E(k) = I_{FC}(k)C(k) = A(1+L(k)+R(k))$$
$$Q_E(k) = Q_{FC}(k)C(k) = A(L(k)-R(k)) \quad (38)$$

which, with the exception of the carrier signal in the I channel, are the desired signals needed to obtain the left and right channel information. The carrier component is removed by a high pass filter. The same digital highpass filter is applied to both channels for two reasons. The first reason is that both channels need to have the same gain and amplitude characteristics to recover the right and left information correctly. The second reason is that the Q channel does have some phase information at baseband which needs to be removed so as not to distort the signal and will be discussed in the next section of this paper, but

$$I_{HPF}(k) = A(L(k)+R(k))$$
$$Q_{HPF}(k) = A(L(k)-R(k)) \quad . \quad (39)$$

Now, by adding and subtracting the I and Q channels, the desired left and right channel signal can be found.

$$I_{HPF}(k) + Q_{HPF}(k) = 2AL(k)$$
$$I_{HPF}(k) - Q_{HPF}(k) = 2AR(k) \quad . \quad (40)$$

VII. Digital Phase Lock Loop

All derivations to this point have ignored the phase error present when demodulating most real signals. A phase lock loop (PLL) is typically used to track and estimate the phase error. A digital phase lock loop (DPLL) was designed for the C-QUAM® receiver and the following derivations include the phase error term. The input signal before demodulation looks like

$$R(k) = A[1+L(k)+R(k)] \cdot cos\left(\frac{w_c}{w_s}(k) + \phi_e(k) + \gamma(k)\right). \quad (41)$$

The numerically controlled oscillator (NCO) output would be:

$$N_I(k) = cos\left(\frac{w_c}{w_s}(k) + \hat{\phi}_e(k)\right), \quad (42)$$

where $\hat{\phi}_e(k)$ is the current estimate of the phase error. The resulting demodulator output signal is

$$R_Q(k) = A[1+L(k)+R(k)] \cdot sin(\gamma(k) + (\phi_e - \hat{\phi}_e)) \quad (43)$$

plus double frequency terms which are filtered out by the low pass decimation filters previously discussed. After multiplying by the estimate of the inverse cosine term

$$Q_E(k) = A[1+L(k)+R(k)]\frac{sin(\gamma(k) + (\phi_e - \hat{\phi}_e))}{cos(\gamma(k) + (\phi_e - \hat{\phi}_e))} \quad (44)$$

or

$$Q_E(k) = A[1+L(k)+R(k)]\,tan(\gamma(k) + (\phi_e - \hat{\phi}_e)), \quad (45)$$

It is straight forward to show that

$$Q_E(k) = A\left[\frac{(L(k)-R(k)) + (1+L(k)+R(k))\,tan(\phi_e - \hat{\phi}_e)}{(1 - tan\gamma tan(tan(\phi_e - \hat{\phi}_e)))}\right] \quad (46)$$

Since $tan(\phi_e - \hat{\phi}_e)$ is assumed small as $(\phi_e - \hat{\phi}_e)$ approaches zero

$$Q_E(k) = A[(L(k)-R(k)) + (1+L(k)+R(k))\,tan(\phi_e - \hat{\phi}_e)] \quad (47)$$

and using the same assumption for additional simplification:

$$Q_E(k) = A[(L(k)-R(k)) + tan(\phi_e - \hat{\phi}_e)]. \quad (48)$$

The highpass filter discussed previously is now used to remove the baseband phase term and a lowpass filter is used to isolate this phase term and remove the left and right channel information:

$$T = tan(\phi_e - \hat{\phi}_e). \quad (49)$$

The new phase error estimate would typically be computed by:

$$\hat{\phi}_e(k+1) = \hat{\phi}_e(k) + (\phi_e - \hat{\phi}_e) \quad (50)$$

However, since it is very processor inefficient on a DSP to perform inverse trigonometric functions, $\phi_e - \hat{\phi}_e$ is never calculated. The input to the loop filter is $tan(\phi_e - \hat{\phi}_e)$ and the output is

$$T_{LF}(k) = \beta T_{LF}(k-1) + (1-\beta)T, \quad (51)$$

which is also in the form of $tan(\phi_e - \hat{\phi}_e)$. Also note that if $x = tan(\phi_e - \hat{\phi}_e)$, then

$$sin(\phi_e - \hat{\phi}_e) = \frac{x}{\sqrt{1+x^2}} \quad (52)$$

and

$$cos(\phi_e - \hat{\phi}_e) = \frac{1}{\sqrt{1+x^2}} \quad (53)$$

so that the sine and cosine of the new estimated phase error can be formed from trigonometric manipulation:

$$sin\hat{\phi}_{enew} = cos(\phi_e - \hat{\phi}_e)\, sin\hat{\phi}_{old} + sin(\phi_e - \hat{\phi}_e)\, cos\hat{\phi}_{old} \quad (54)$$

and

$$cos\hat{\phi}_{enew} = cos(\phi_e - \hat{\phi}_e)\, cos\hat{\phi}_{old} + sin(\phi_e - \hat{\phi}_e)\, sin\hat{\phi}_{old} \,. \quad (55)$$

Since $\frac{w_c}{w_s} = \frac{1}{4}$, only four values of sine and cosine need to be stored in a modular table to compute the NCO outputs which are then updated with the new phase error estimate using the same trigonometric manipulations used in equations (44) and (45). The final NCO output values are

$$N_I(k) = cos\left(\frac{w_c}{w_s}(k) + \hat{\phi}_e\right) \quad (56)$$

and

$$N_Q(k) = sin\left(\frac{w_c}{w_s}(k) + \hat{\phi}_e\right), \quad (57)$$

which can be calculated using trigonometric identities in terms of $sin\hat{\phi}_e$ and $cos\hat{\phi}_e$ as shown in the previous equations. Note that finding the value of $\hat{\phi}_e$ is never required using this expansion.

VIII. VERIFICATION

The test system used to develop and verify the AM stereo is shown in Figure 8. The Leader AM Stereo Signal Generator is the heart of the test system and is designed to generate C-QUAM® signals from baseband inputs. A CD player is used to generate the left and right signals used as input to the generator. The output of the generator is the fed to the input of an AM receiver front end chip which performs the tuning and downconversion of the signal to 450 Khz.

As described in section 3.0, the next stage does an additional downconversion and samples the signal for input into the DSP development system. The signal is then processed and the left and right channel information is output to an amplifier speaker combination. The development system is controlled by a host computer to allow for efficient development and testing of the design.

IX. CONCLUSION

The ability to incorporate all functions as firmware solutions is critical in designing DSP based audio systems. Not only has this paper presented a method for digital AM radio reception, the resulting signal is superior to the signal of the analog circuitry typically used. An expensive analog receiver available today typically produces 40 dB of channel separation. The digital implementation discussed in this paper achieves more than 40 dB of channel separation and is just the initial implementation. Additional enhancements which are anticipated to improve the signal will be implemented and verified. The DSP implementation provides a flexible and higher fidelity solution for AM radio reception.

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A DIGITAL APPROACH TO AN FM EXCITER

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ABSTRACT

This paper will discuss the advantages of digital FM modulation relative to traditional analog techniques. The Direct Digital Frequency Modulation (DDFM) is centered around a Direct Digital Synthesizer (DDS) which allows true digital generation of the FM signal. This digital technique has achieved superior phase noise characteristics, linearity, frequency resolution, and the ability to select any FM frequency without adjustment. (See Figure 1). The design approach allows the digital exciter to accept existing analog inputs or can interface to digital sources of the future including a stereo generator.

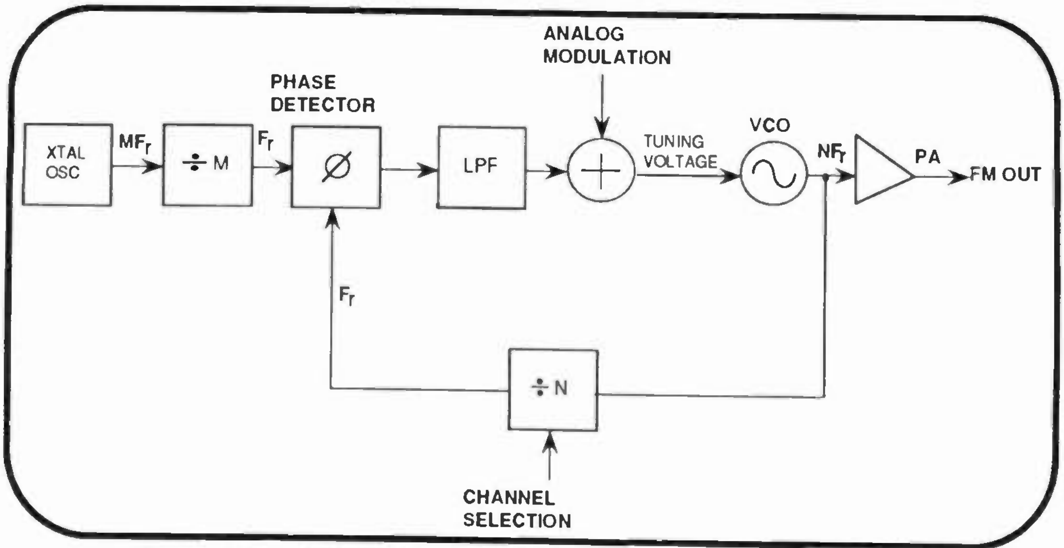
Amplitude Response	+0.01dB (20Hz-100kHz)
Phase Response	+0.03 (20Hz-53kHz) +0.1 (20Hz-100kHz)
FM S/N	95dB
THD	0.003%
SMPTE	0.0045%
CCIF	96dB
DIM	0.006% (With 100kHz Pre-Filtering)
Synchronous AM	50dB
Asynchronous AM	60dB
Stereo Separation	60dB
Stereo S/N	80dB (Current test equipment limited to 80dB)
Stereo THD	0.03% (Current test equipment limited to 3%)
RF Spurious	90dB (Outside of + 600kHz FCC window, 80dB Inside.)

TYPICAL PERFORMANCE SPECIFICATIONS BELIEVED TO BE ACHIEVABLE FOR PRODUCTION LIMITS
Figure 1

Traditional FM exciter technology utilizes a modulated oscillator for FM generation. This simple and low-cost approach of modulating an oscillator to move proportionally to a voltage has been used for the last two decades. Usually it is accomplished by applying a modulation signal to a varactor which is part of an oscillator's tank circuit. The tank circuit's center frequency changes as the modulation voltage changes. As the center frequency changes, the frequency of oscillation also changes.

Figure 2 shows a basic block diagram of a traditional modulated oscillator. A crystal oscillator is divided down based on PLL design criteria. The minimum frequency increment of the modulated oscillator of most analog FM exciters is typically 10 kHz steps. This reference frequency is divided by a channel selection factor to obtain the desired FM channel. Therefore, channel selection is made by changing the channel selection component or possibly the 10 kHz step. The Voltage Controlled Oscillator also receives an error voltage from the phase detector, which keeps the oscillator on frequency should it start to drift.

Modulation is applied at the VCO input, changing the frequency proportionally. This change in frequency is sensed by the phase detector, which produces a voltage equal to the modulation, but opposite in phase. This signal must be filtered by the low pass filter to keep the modulation from being cancelled.



ANALOG FM MODULATION OSCILLATOR BLOCK DIAGRAM
FIGURE 2

Therefore, the low pass filter attenuates all modulation frequencies. This requires a cutoff frequency well below 20 Hz.

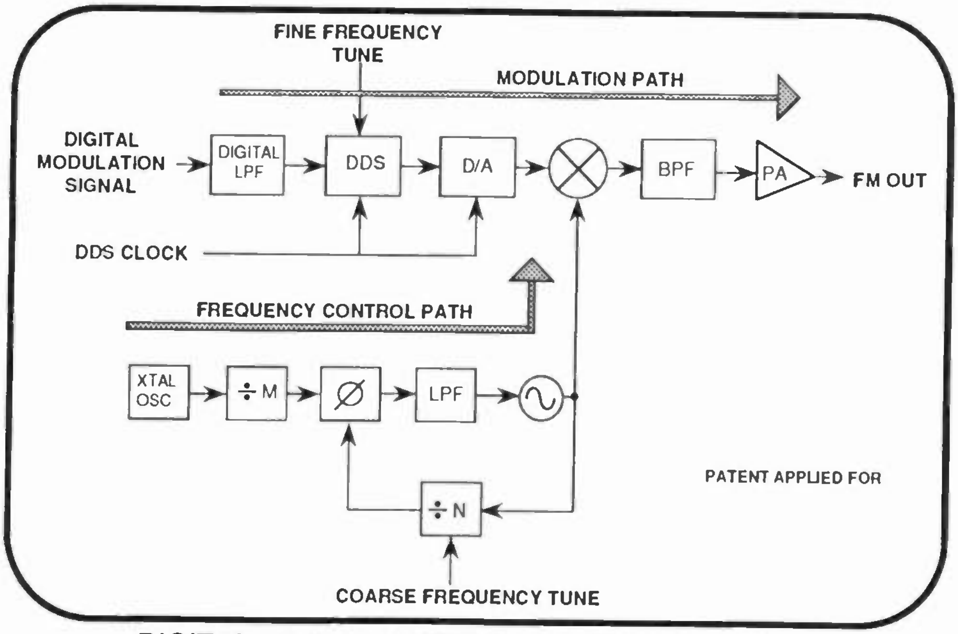
While this is essential for modulation, it provides little noise reduction and inhibits fast loop response. Slow loop response is usually overcome by using a faster loop to obtain lock, then switching to the narrow band loop. The modulated oscillator phase noise is therefore only slightly better than the phase noise of the VCO. Also, because lower frequencies are attenuated less, the low frequency audio response is degraded.

Another problem due to the narrow bandwidth of the modulation oscillator is susceptibility to microphonics. Unless adequate mechanical isolation is provided, microphonic frequencies above a few Hertz will degrade system noise performance. Additionally, system performance is impacted by the non-linearities in the VCO itself. These factors limit the performance of an analog exciter, and led to the development of a digital approach to an FM exciter which overcomes them.

DIRECT DIGITAL FM TECHNIQUE

With the availability of single chip, low-cost Direct Digital Synthesis (DDS) devices, digital frequency modulation can be used, eliminating many of the analog disadvantages. DDS technology also allows direct interface to new digital stereo and other sources that are beginning to be developed. This is one of the many motivations behind the development of a true digital FM exciter. In a DDS system, FM generation is accomplished discretely. Figure 3 shows a block of a DDS device in the exciter or Numerically Controlled Oscillator (NCO) as they are sometimes called. The frequency tuning of the DDS is similar to that of an analog VCO. Most DDS devices have an frequency adder which allows a digital off set adjustment of the center frequency. This offset can be compared to the modulation input of the VCO in an analog exciter.

The DDS adder input can be from any digital source such as a digital stereo generator. Digital tuning is then accumulated and the result is a number



DIGITAL FM MODULATION BLOCK DIAGRAM
FIGURE 3

representing the instantaneous phase of the FM signal. The sum of the center frequency and the DDS adder input to the phase accumulator will eventually overflow. This overflow of the accumulator simply means the phase has completed one revolution.

DDS technology typically offers 32-bit performance for both the modulation and frequency inputs guaranteeing a frequency resolution of less than 0.005 Hz using a 20 MHz clock. Current semiconductor IC capabilities force the accumulated phase to be truncated to 16 bits, typically. The truncating phase bits generate spurs, and for a 16 bit sine look-up table, the maximum spur level due to phase quantization is measured to be approximately 96 dB.

The FM output signal will exhibit improved phase noise performance over the DDS clock. The most difficult problem to this Direct Digital approach has been the Digital to Analog (DAC) process. With current DAC technology limited to 12-bit resolution and below for these speeds, the phase quantization errors of the DAC are much larger than those of the DDS device.
However, spur locations and amplitudes can be predicted, allowing for intelligent selection of output frequencies. The largest spurs are due to harmonics and aliases, or image products. Therefore, a practical system requires intelligent selection of clock frequency and output frequency to minimize high order and low order spurs. A bandpass filter was designed to attenuate the spurious products. This filter features very flat amplitude and group delay characteristics so the perfectly linear digital modulation advantages are not lost. This filter which has been found to have the largest impact on system linearity and thus, performance.

DIGITAL MODULATION SOURCE

Several requirements must be met by the digital modulation signal to make a DDS system feasible. First, the data must have a

DIGITAL MODULATION SOURCE

Several requirements must be met by the digital modulation signal to make a DDS system feasible. First, the data must have a sample rate that is coherent with the clock. This sets a requirement that the modulator be slaved to the digital stereo source. This is easily done if an Analog to Digital (A/D) convertor is used. In this mode, the A/D and the DDS device operate from the same clock. If the source is a digital stereo generator operating from another clock, the DDS must be phase-locked to the stereo generator. Second, the amplitude resolution should be CD quality, allowing transparent operation relative to the digital source. This requires 16 bits or more of resolution. Third, the sample rate should be compatible with system frequency selections, so modulation sidebands do not distort the desired FM channel.

UP CONVERSION SCHEME

Due to current DAC limitations, the output frequency of a digital modulator is well below the FM band. Up-conversion of the FM signal is therefore required. Because the modulation path is separate from the frequency control path, the PLL (Phase Lock Loop) bandwidth can remain wide. This eliminates the problem associated with the narrow loop bandwidth of the analog modulation oscillator.

VCO (Voltage Controlled Oscillator) linearity is no longer a concern because modulation does not pass through the loop. FM deviation is always the same from channel to channel. This feature makes the system ideal for frequency agile systems such as N+1 applications. The coarse frequency tuning on the PLL provides a convenient place for channel selection. Fine tuning of the frequency can be obtained by the DDS device itself, thus providing selectivity far superior to the 10 kHz provided in most analog exciters. The coarse tuning resolution needs only to be equal to the maximum movement allowed in the DDS fine tuning. The noise of the up-convertor will dominate the overall system noise, because the digital modulator has exceptional phase noise characteristics.

Digital Exciter System Performance

Actual tests have been made to verify advantages of a digital approach to an FM exciter. System linearity was found to be substantially better than analog capabilities. Careful attention was paid to specified DAC linearity, and more importantly, the design of the alias suppression filter. The phase noise was limited, as expected, by the up-convertor, but FM S/N levels better than 95 dB were achieved using de-emphasis. Deviation and Linearity stayed relatively constant throughout the FM band. Channel selection could be done at a very fast speed (kHz). Output frequency resolution was well below the crystal oscillator drift of 300 Hz. Over a limited bandwidth that allows reasonable PLL frequency steps, the spurious levels were below 90 dB.

CONCLUSION

A digital approach to FM modulation has been shown to be a viable improvement to the traditional analog exciter. Significant performance improvements were achieved in key areas relative to FM generation. The technology now exists for a totally digital link between the source and FM generation. The digital exciter design features a flexible approach to the various input signals.

SELECTION OF FM ANTENNA ELEVATION PATTERNS

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Summary

The elevation radiation pattern of an FM broadcast transmitting antenna is an important factor in the performance of the FM broadcast system. The elevation pattern determines antenna gain and electromagnetic energy exposure levels at ground level and impacts the reliability of reception at locations near the transmitter site. The optimal number of FM antenna elements and their spacing may be determined based on reasonable engineering considerations, taking into account the elevation of the antenna above the population to be served, the distance from the transmitter site to the closest significantly populated areas, energy exposure considerations at or near the transmitter site, and the minimization of antenna-induced multipath distortion. A better definition of engineering objectives and solutions improves the precision of the cost/benefit analysis that accompanies any purchase decision.

Introduction

When a particular FM transmitting antenna bay count and interbay spacing combination is chosen, the characteristics of a directional pattern are being selected. However, this directivity is in the elevation, not azimuthal, plane, an axis perpendicular to the service area usually worried about in connection with directional arrays. What characteristics of the elevation pattern are important? How do they impact a station's service? What antenna parameters determine the elevation pattern? From a better understanding of the nature of radiating characteristics, we may obtain insight into FM antenna selection based more on true engineering considerations and less on salesmen's recommendations and guidance of similar reliability. Such insight will help the station or consulting engineer give the antenna manufacturer requirements that encourage delivery of a product which will contribute to optimal service, minimized radiation hazard exposure, and good operating efficiency.

Conventional Wisdom

One of the current American cultural buzzwords is "conventional wisdom", or "CW". What does CW say about antenna selection? An informal survey of several engineers experienced in FM transmitting facility configuration was undertaken, with the results summarized below. ERP is effective radiated power and HAAT is antenna height above average terrain.

Table I

ERP (kW)	HAAT (ft)	Bays	Xmtr (kW)
3	300	3	2.5
6	328	4	3.5
25	328	4	11
50	500	5	20
100	1000	8	30

Some readers of this paper will have different opinions, of course. Such is the nature of this topic.

System Cost

Another popular criterion for use in FM antenna selection is cost: what is the least expensive transmitter/antenna combination? Application of typical cost data yields the result shown next.

Table II

ERP (kW)	HAAT (ft)	Bays	Xmtr (kW)
3	300	3	2.5
6	328	4	3.5
25	328	5	10
50	500	6	20
100	1000	10	25

Not surprisingly, Tables I and II have similarities. Dollars usually have a way of influencing engineering decisions. In this paper, antenna selections based on cost concerns will be identified by the acronym "CC".

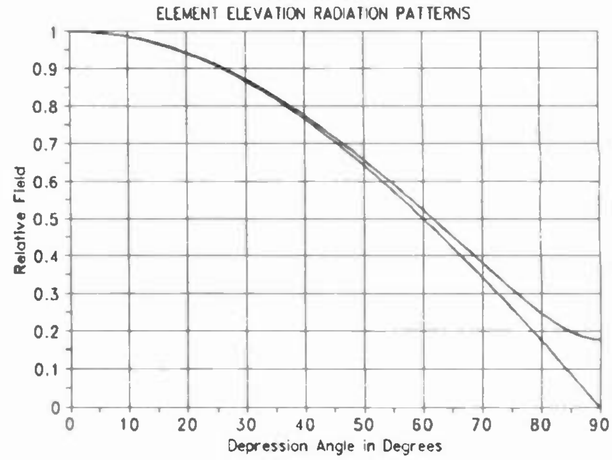
FM Antenna Basics

VHF transmitting antennas, be they used for FM or television service, are, fundamentally, directive arrays ... that is, directional in the elevation plane. The elements are usually spaced, phased, and powered uniformly (or at least that is what is often intended). Change or imbalance of any or all of the foregoing quantities will change the antenna's elevation pattern.

The elevation radiation pattern of an FM broadcast antenna is actually the product of two patterns: one for the radiating element used and the other for the array that such elements will be inserted into. While element patterns are unique to each manufacturer's design, array factor patterns are defined by mathematics which are common all radiators.

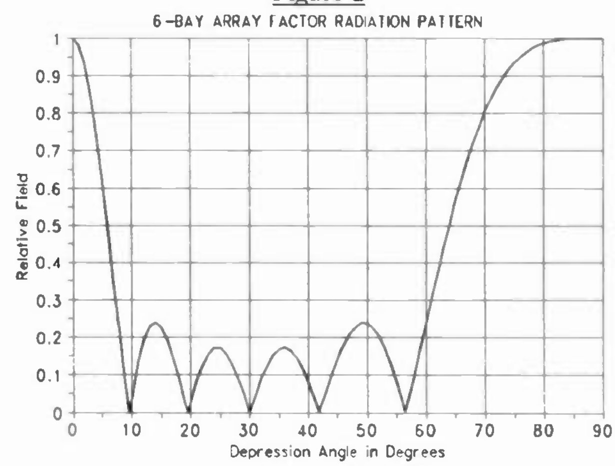
Figure 1 illustrates two single-element patterns. The lower curve is for a pure cosine function, the theoretical pattern of a horizontally polarized ring antenna. The upper curve is the cosine function with null fill added in quadrature. The fill parameters are selected such that the minimum radiation is no less than 15 dB below the maximum. This "generic" element pattern will be presumed throughout the remainder of this paper, rather than using any particular manufacturer's radiating element.

Figure 1



The array factor elevation pattern shown by Figure 2 represents that resulting from a 6-bay, full wavelength spaced antenna. Note that the broadest

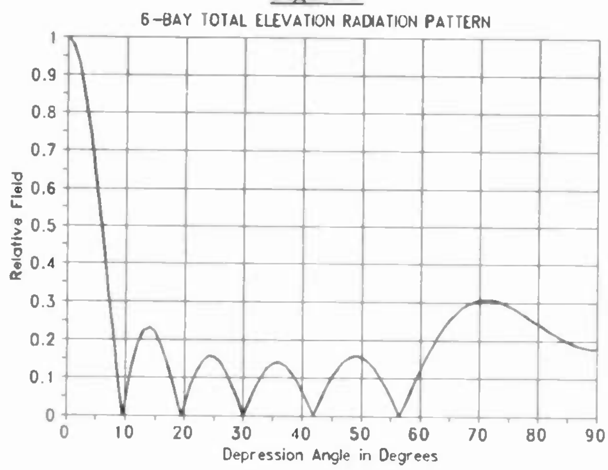
Figure 2



beam of the array factor elevation pattern is directed upward and downward, i.e., toward 90 degrees. The horizontal lobe is much smaller. Were it not for the limited downward radiation of the element pattern, most radiated energy would be directed up and down the tower, not outward and toward the audience!

The complete elevation pattern, the product of Figures 1 and 2, is shown by Figure 3. The impact of the element pattern in reducing the downward (grating) lobe is quite obvious. If the array factor pattern did not produce such a strong lobe at or near 90 degrees, it would not be necessary to rely completely on the element pattern for minimization of downward/-upward radiation.

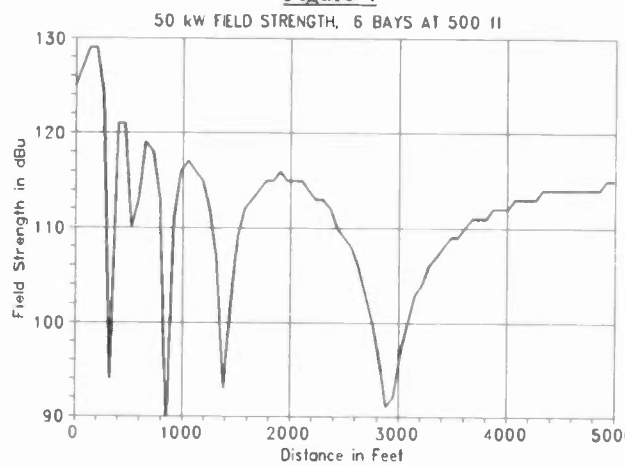
Figure 3



The elevation pattern will affect station coverage, to at least a limited extent. The array gain toward the radio horizon, which is virtually perpendicular to the axis of the antenna, will determine the effective radiated power, and thus coverage reach, achieved. The radiation characteristic at downward angles will determine the presence or absence of field strength "scalloping" close to the antenna, with its attendant antenna-induced multipath distortion zones, and rings of "blanketing" receiver overload interference in the same area. The smaller the "scalloping" zone, the fewer its minima, and the less the difference between adjacent maxima and minima, the fewer the problems will be encountered in close-in service.

Figure 4 illustrates the calculated field strength close to an FM station operating with an effective radiated power (ERP) of 50 kW into a conventional 6-bay, full wavelength spaced antenna having a radiation center elevation of 500 feet above flat ground. Such a configuration comports with both CW and CC. As can be seen, field strength minima and radial "scalloping" occur out to a distance of about 0.8 mile (1.3 km). Field strength at radiation maxima varies by 14 dB; variations including minima are even worse. Therefore, reception quality will be more inconsistent than usual within that distance from the antenna system, because of the elevation pattern minima and sidelobe effects. This region will be referred to as the *antenna-induced multipath zone* herafter in this paper.

Figure 4



The antenna characteristics that are of particular importance in determining the antenna-induced multipath zone are the depression angle to the first elevation pattern minimum off the horizon and the radiation within that minimum, usually referred to as *null fill*. The smaller this depression angle, the farther from the transmitter the "scalloping" zone of inconsistent reception quality will reach. The greater the fill of the elevation pattern minima, the less pronounced reception anomalies will be.

Bay/Spacing Product

Maximum acceptable reception inconsistency radii may be defined which may be translated into depression angles and, ultimately, into FM antenna selection criteria. The depression angle of the first minimum in the elevation pattern is readily calculated using the following equation:

$$\Theta = 90 - \cos^{-1}\left(\frac{1}{nd}\right) \qquad \text{Eq. [1]}$$

where: Θ = depression angle in degrees
n = number of bays
d = interbay spacing in wavelengths.

This equation is presented in several well-known antenna textbooks.[3] At conventional full wavelength spacing, the depression angles for common bay counts are detailed by the following data.

Table III

Number of Bays	Depression Angle (deg)
2	30.0
3	19.5
4	14.5
5	11.5
6	9.6
7	8.2
8	7.2
10	5.7
12	4.8

Take particular note of the fact that the denominator of the depression angle equation is the product of bay count *and* interbay spacing. A depression angle of 9.6° may be obtained from 6 bays and 1λ spacing, 7 bays and 6/7λ spacing, 8 bays and 3/4λ spacing, etc. The denominator of Equation 1 will be cited hereafter within this paper as the *bay/spacing product*.

The distance to the first pattern minimum is dependent upon the antenna height above terrain near that minimum and the depression angle. The following simple equation yields that distance:

$$d = \frac{h}{\tan \Theta} \qquad \text{Eq. [2]}$$

where: d = distance in same units as h
h = antenna radiation center height above terrain in first minimum
Θ = depression angle to first minimum

In rolling terrain, iterative solution of this equation might be necessary. However, for the simplistic purpose of this paper, flat terrain will be assumed. Tables IV and V, presented at the end of the text of this document, show first minima distances for various bay/spacing products and antenna elevations and the percentage of the potential service area that such distance represent. Where the spacing is equal to one wavelength, the bay/spacing product is simply the bay count.

Placement of the First Null

The tolerable distance to the first null is not obvious simply by looking at Tables IV and V. Meaningful criteria need to be developed for guidance in selecting a first null distance. The matter becomes considerably more complicated when station owners or managers reside in the immediate vicinity of the antenna site!

The first reasonably simple criterion for determining null placement is based on the idea that the area subject to potential multipath distortion, induced by the radiating characteristic of the transmitting antenna, should not exceed a minimal percentage of the station's total service area. The nominal service area of the station is defined. A null zone radius is determined by multiplying the square root of the permissible multipath area proportion by the nominal service radius. From this radius and the antenna height, the first null depression angle may be found.

The first example of application of this technique is based on the service radii for various FM station classes defined by the FCC. Depression angles to the first null are calculated using the FCC-defined maximum protected service radii and several antenna heights. Where antenna heights are below the nominal maxima for the station class, service radii were determined as the distance to the protected service contour for the maximum ERP permitted for that class and the specified antenna height. In Table VI, the antenna-induced multipath zone is limited to 0.1 percent of the FCC-defined service radius. The bay/spacing product was computed from the depression angle using Equation 2, truncating the result, and subtracting one unit.

Table VI

Depression Angles From FCC Service Radii

Station Class	Service Radius (mi)	Antenna Height (ft)	Depression Angle (deg)	Bay/Spacing Product
A3	11½	200	6.0	8
	15	300	6.9	7
	"	400	9.2	5
	"	500	11.6	3
A6	14	200	4.9	10
	17½	300	5.9	8
	"	400	7.9	6
	"	500	9.9	4
C3	19½	200	3.5	15
	24½	300	4.2	12
	"	400	5.6	9
	"	500	7.0	7
B1	23	200	3.0	18
	27½	300	3.7	14
	"	400	5.0	10
	"	500	6.3	8
C2	27	300	3.8	14
	32½	500	5.3	9
	"	700	7.4	6
	"	1000	10.7	4
B	34½	300	3.0	18
	40½	500	4.2	12
	"	700	6.0	8
	"	1000	8.5	5
C1	36½	500	4.7	11
	40½	700	6.0	8
	45	1000	7.7	6
	45	1500	11.6	3
C	45	1000	7.7	6
	48	1200	8.6	5
	52	1500	10.0	4
	57	2000	12.2	3

The truncation and unitary subtraction adjustments assure that the first null will fall short of the 0.1 percent service area radius, preserving reliable service beyond that distance. More bays can be used with decreased spacing to keep the bay/spacing product within the limitations shown. The technically sophisticated reader is advised to compute the bay-spacing product directly from the depression angle using Equation 1 and make truncation/ subtraction adjustments based on his or her knowledge.

Table VI was constructed based on FCC definition of service area radii. In the absence of interference and with proper selection of ERP, meaningful service may be realized to the radio horizon that is determined by the station's antenna height.[4] Table VII is based solely on antenna heights and the service radii that result therefrom. Neither the FCC's class definitions nor ERP levels are relevant to this table, because it is presumed that the station operates with enough ERP to achieve usable service at the horizon.

Table VII

Depression Angle From Horizon Radius

Antenna Height (feet)	Horizon Radius (miles)	Depression Angle (deg)	Bay/Spacing Product
150	17½	3.0	18
200	20	3.4	15
300	24½	4.2	12
400	28½	4.8	10
500	31½	5.4	9
600	34½	5.9	8
800	40	6.8	7
1000	44½	7.6	6
1200	49	8.4	5
1500	55	9.3	5
2000	63	10.7	4

Based on the criteria that underlie the construction of Table VI, stations operating at the maximum ERP/HAAT for their respective classes should not exceed the following wavelength-spaced counts: 3 kW Class A: 7, 6 kW Class A: 8, Class C3: 12, Class B1: 14, Class C2: 9, Class B: 12, Class C1: 8, and Class C: 3. However, few Class C stations achieve the maximum permitted HAAT. For typical Class C stations, maximum bay counts of 4 to 6 are shown.

These results may surprise some readers. For the lower classes of stations and those using shorter antenna heights, the data suggest that many more bays than the counts presumed under CW or CC may be used. However, for Class C stations, the bay count is substantially less than what most people would presume and the facilities that most such stations have. Fortunately, the antenna height required for Class C stations usually dictates a site location that is well separated from densely populated areas, so a larger radius of antenna-induced multipath distortion is more acceptable than for most other facilities.

The prudent station engineer or consultant should go beyond the simplistic area/percentage criteria presumed in Tables VI and VII in determining optimal depression angle and bay/spacing product. The depression angle may be lowered and bay/spacing product increased if the population density near the antenna site is low. Conversely, the opposite conclusion might be reached if the population density is high, such as is the case for Class B stations located in the center of urban areas. Population counts and/or references to current maps of the transmitter site area may be used to define the antenna-induced multipath zone radius and, from that, depression angle and bay/spacing product on a more meaningful basis.

Of course, null fill can mitigate most of the antenna-induced multipath distortion in first null region. However, positioning of the first null is important, regardless of fill, because it is a convenient way of establishing the outer boundary for careful analysis of antenna-induced field strength variation with distance. Furthermore, it provides yet another guidepost for use in establishing the bay count and interelement spacing parameters.

The foregoing analysis may be used as a guide in selecting the optimal number of wavelength-spaced bays for a particular station situation. However, it is more elegantly used to establish a bay/spacing product, with the particular bay count and spacing parameters to be established based on additional considerations.

Array Factor Pattern

The specific number of bays and spacing thereof define the array factor elevation pattern which, as described earlier in this paper, is multiplied by the individual radiating element pattern to produce the complete elevation pattern for the antenna array.

A quantity of substantial interest in defining the array factor pattern is the downward, or grating, lobe that is produced. This lobe usually has the most significant effect on electromagnetic energy levels on the tower and/or near its base. In the interest of safety for workers and the public, this lobe should be minimized. Figure 5 shows the magnitude of the array factor grating lobe, as a function of interbay spacing, for 2 to 5 bays, while Figure 6 illustrates the same dependence for 6 to 12 bays. For graphical clarity, bay counts of 7, 9, and 11 have not been shown.

Figure 5

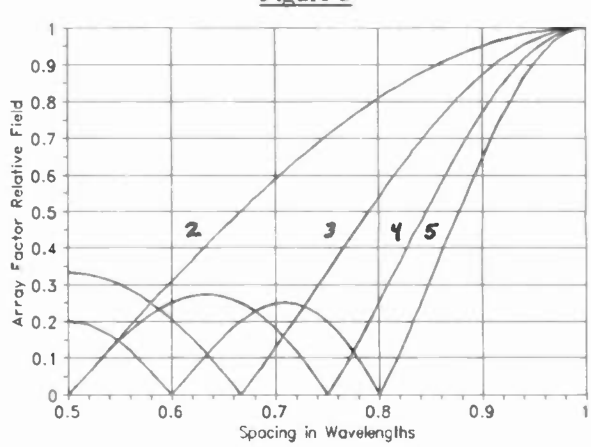
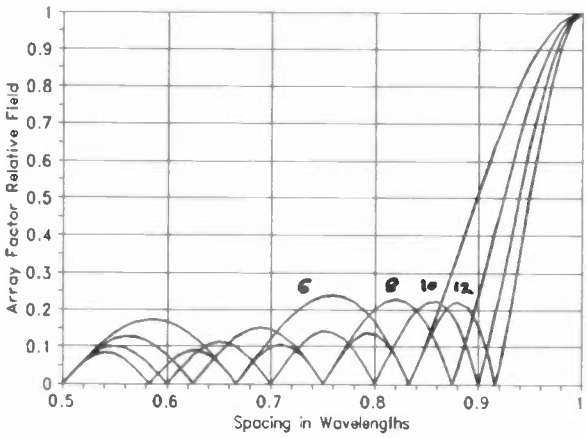


Figure 6



The normalized array factor pattern for an array having identical element fields and phases is defined by Equation 3:[5]

$$f(\Theta) = \frac{\sin[180nd\ \cos(90-\Theta)]}{n\ \sin[180d\ \cos(90-\Theta)]} \quad \text{Eq. [3]}$$

where: n = number of bays
d = interbay spacing in wavelengths
Θ = depression angle in degrees

The array factor equals zero whenever its numerator goes to zero but its denominator does not. At a depression angle of 90 degrees, this will occur whenever the spacing is *not* exactly one wavelength and the bay/spacing product equals a multiple of one wavelength. In other words, it is mathematically possible to create a zero array factor with *any* reasonable inter-element spacing *other* than one wavelength ... as long as enough bays are used. More practically, it is possible to create a zero downward array factor for any number of elements greater than one, with multiple zeroes achievable for antennas having four or more elements.

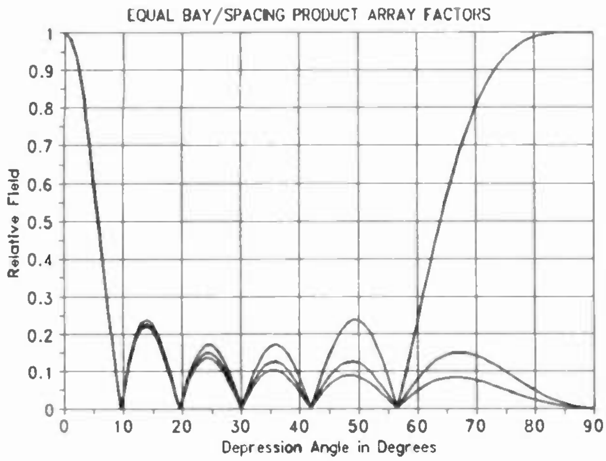
Inspection of Figures 5 and 6 reveals that, contrary to popular myth, a half wavelength spaced antenna is *not* the only configuration that will minimize grating lobes. The maximum spacings at which grating lobe minima can be obtained are 0.75λ for 4 bays, 0.8λ for 5 bays, 0.833λ for 6 bays, 0.875λ for 8 bays, 0.9λ for 10 bays, and 0.917λ for 12 bays. If these spacings seem familiar, that is because each one equals one less than the number of bays, divided by the number of bays. What could be simpler? Minimization of downward radiation does *not* necessarily require use of a half-wavelength spaced array.

Another interesting fact is that each bay/spacing product that produces a zero downward array factor also produces a depression angle to the first null equalling that resulting for an integer number of wavelength-spaced bays. In other words, for most numbers of wavelength-spaced bays, an equivalent main lobe can be developed coincidental with a downward radiation minimum, for several alternative bay/spacing combinations.

Main Lobe Equivalence

Equation 1 shows that it is the bay/spacing product that determines the depression angle. The same depression angle is obtained using 6 bays at 1λ spacing, 8 bays at 3/4λ spacing, 10 bays at 3/5λ spacing, and 12 bays at 1/2λ spacing, etc. It may be inferred that matching the depression angle to the first null will match array factor elevation patterns, at least in that region, as well. Figure 7 shows the array factor elevation patterns for the 6, 8, and 12 bay combinations described in this paragraph. The uppermost plot is for 6 bays; the center plot shows an 8 bay pattern, and the lower plot corresponds to 12 bays. These elevation patterns are essentially identical to the second null. The number of maxima and minima are identical for all three bay/spacing combinations. Significant differences in elevation pattern maxima are found as the interbay spacing decreases and depression angle

Figure 7

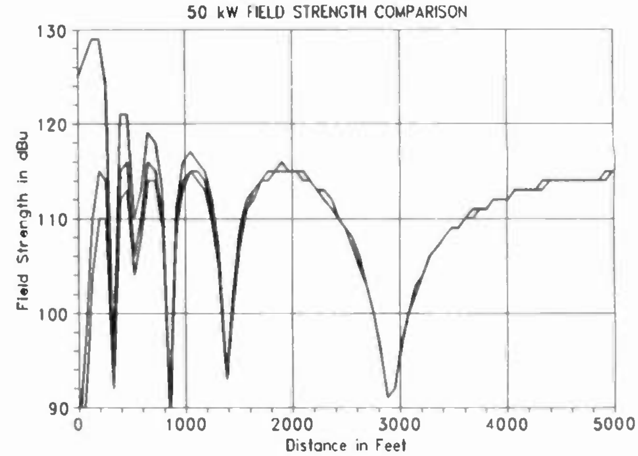


increases. The greater the number of bays and smaller their spacing, the lower the radiation at large depression angles, i.e., near the antenna site. For a given bay/spacing product, the differences between elevation patterns as bay counts are varied lie not in the main lobe shape, but in the magnitude of sidelobe maxima at large depression angles. All of these differences arise from the array factor elevation pattern.

The severity of antenna-induced multipath can be limited somewhat if the sidelobes of the array factor pattern are balanced to produce similar field strength maxima along the ground. Presuming flat terrain, this is simply a matter of finding the bay count and spacing that simultaneously achieve a grating lobe minimum, a desirable gain figure, and uniform sidelobe field strength peaks.

Figure 8 illustrates the field strength versus distance relationship, presuming flat terrain, for 6 bays at full wavelength spacing, 8 bays at 3/4 wavelength spacing, and 12 bays at half wavelength spacing. The highest field strength peaks result from the 6 bay configuration and the lowest correspond to the 12 bay antenna.

Figure 8



Note that the number and position of field strength maxima and minima is the same for all such antennas. The difference is the magnitude of the field strength peaks that result from the sidelobes. At 8 bays or more, peaks are reasonably uniform.

Power Gain

Many antenna configuration decisions are made on the basis of power gain, particularly when cost of the transmitter/line/antenna combination is of primary importance. Gain is affected by the number of bays and the spacing between them. Primarily due to a lack of generally available information regarding gain/spacing relationships, there is often reluctance by station engineers to consider antennas having less than full wavelength spacing.

Figures 9 and 10 show power gain versus interbay spacing for arrays of 2 to 5 and 6 to 12 bays, respectively, based on the "filled cosine" element pattern of Figure 1. Results will vary somewhat with other elements being used, but the relationships shown are typical of most modern designs from reputable manufacturers. Note that gain is maximum at spacings just below one wavelength.

Figure 9

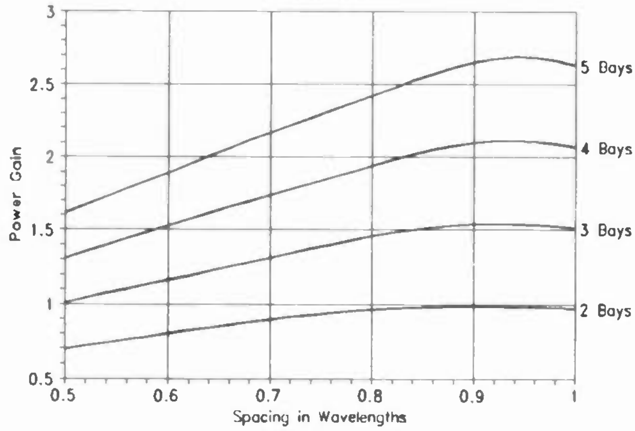
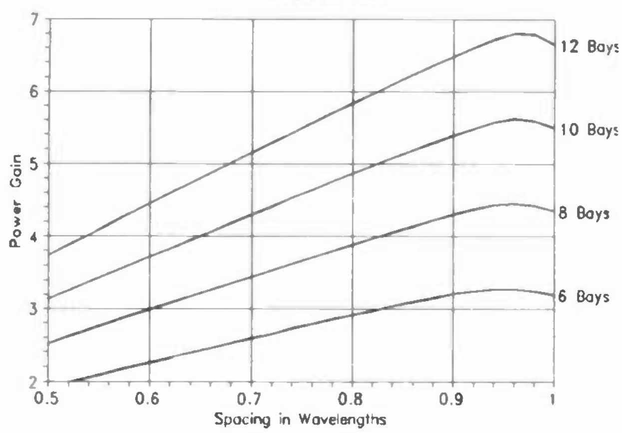


Figure 10



The same gain achieved at full wavelength spacing is also achieved at a slightly lesser spacing, with a substantial reduction of the grating (downward) lobe of the array factor pattern, as detailed in the following Table VIII. Keep in mind the fact that downward power density, the quantity used in evaluating electro-

Table VIII

Gain Equivalence at Reduced Spacing

Number of Bays	Spacing	Array Factor
2	0.82	0.844
3	0.85	0.725
4	0.87	0.628
5	0.88	0.647
6	0.90	0.513
8	0.91	0.345
10	0.92	0.236
12	0.93	0.042

magnetic radiation exposure levels, is proportional to the *square* of the array radiation factor, so substantial reduction of downward radiation may be obtained with the same or greater gain.

Antenna Equivalence

The fact that certain antenna configurations produce equivalent depression angles to the first null and identical major lobe and first sidelobe radiation was noted earlier in this paper. What are the aperture, gain, and energy exposure tradeoffs of these different implementations of the same theme? Table IX provides such data for different bay/spacing combinations, grouped to generate equivalent main lobes. Energy exposure calculations are based on 100 kilowatts, circularly polarized, at 328 feet (100 meters) above ground level and represent the worst case value anywhere along flat ground in the vicinity of the tower.

Note that the first or second step of reduced spacing causes the greatest increase in gain and decrease of electromagnetic energy exposure. Because a noticeable gain increase is obtained from a slight increase in aperture, it may be possible to hold bay count constant, reduce spacing, increase depression angle by one step, reduce energy exposure, and continue to use the existing transmitter, in some cases.

Practical Limitations

FM antennas having interbay spacings not equal to one-half or a full wavelength are rare, particularly for side-mounted antennas, the most popular type. This arises largely from manufacturing considerations. For maximum gain, each FM radiating element is fed in phase. It is relatively easy to achieve in-phase feeds for interbay spacings of one-half or a full wavelength using simple series connections with "tee" junctions for each side-mounted element on a common feed line.

Table IX

Equivalent Antenna Parameters

Depression Angle (deg)	**Number of Bays**	**Interbay Spacing** (λ)	**Array Aperture** (λ)	**Gain**	**Energy Exposure** (μW/cm²)
30.0	2	1.000	1.00	0.97	120
"	3	0.667	1.33	1.26	19.8
"	4	0.500	1.50	1.31	17.4
19.5	3	1.000	2.00	1.51	92.6
"	4	0.750	2.25	1.84	11.6
"	5	0.600	2.40	1.89	8.1
"	6	0.500	2.50	1.91	7.8
14.5	4	1.000	3.00	2.07	76.9
"	5	0.800	3.20	2.42	8.4
"	6	0.667	3.33	2.48	5.1
"	7	0.571	3.43	2.51	4.5
"	8	0.500	3.50	2.52	4.4
11.5	5	1.000	4.00	2.63	66.6
"	6	0.833	4.17	3.01	6.8
"	7	0.714	4.29	3.07	3.7
"	8	0.625	4.37	3.10	3.0
"	9	0.556	4.44	3.12	2.9
"	10	0.500	4.50	3.13	2.8
9.6	6	1.000	5.00	3.20	59.4
"	7	0.857	5.14	3.60	5.8
"	8	0.750	5.25	3.67	2.9
"	9	0.667	5.33	3.70	2.3
"	10	0.600	5.40	3.72	2.0
"	11	0.545	5.45	3.73	2.0
"	12	0.500	5.50	3.74	2.0
7.2	8	1.000	7.00	4.35	49.9
"	9	0.889	7.11	4.79	4.6
"	10	0.800	7.20	4.86	2.1
"	11	0.727	7.27	4.89	1.5
"	12	0.667	7.33	4.92	1.3
5.7	10	1.000	9.00	5.50	44.0
"	11	0.909	9.09	5.98	3.9
"	12	0.833	9.17	6.06	1.7

Other separations, particularly those where the interbay spacing is less than 0.8λ, are not as easily fed. It may be necessary to utilize a power divider located at the antenna center and equal length feed lines to each bay in order to achieve proper phasing, as is done in panel antennas. The feed system may require custom design and certainly involves more cost than the simple arrangements used with one-half and full wavelength spacings.

The incremental cost increase inherent in a more sophisticated feed system should be weighed against the advantages obtained therefrom. Minimization of downward radiation and its non-uniformity may help control not only antenna-induced multipath distortion, but also reduce the magnitude of sidelobe reflections off surfaces near the antenna site, lessening other multipath problems, and reducing the potential for blanketing interference. The incremental cost increase is not overly significant in the context of a complete transmitter, line, and antenna installation. And the cost of the FM antenna is still far less than that of a typical AM station directional antenna that garnered the same audience rating percentage during that medium's heyday 25 years ago. Antenna configuration is an area where the primary motivation should not be the minimization of expense.

Also keep in mind that the structure which supports a side-mounted antenna will have some impact on the elevation pattern achieved in actual practice. This will be particularly true when tower member and member loop dimensions approach resonant lengths. Care should be taken in antenna support and mounting to minimize the likelihood of antenna-induced multipath distortion caused by interaction between the antenna and the structure to which it is attached. Of course, panel antennas are generally unaffected by tower dimensions, particularly when they are properly installed.

In the interest of space, time, and simplicity, this paper has not considered adjustment of element powers to fill one or more nulls of the elevation pattern. For stations located in urban areas, null fill may be an important consideration in antenna design. The station or consulting engineer facing such a situation should engage antenna manufacturers in a serious discussion of this topic.

Conclusion

Selection of FM antenna elevation patterns can be made on the basis of sound engineering considerations. The bay/spacing product should be selected to control the location of the first elevation pattern null. The number of bays and their spacing should be configured to achieve efficient gain, minimum downward radiation, and approximate uniformity of field strength peaks within the antenna-induced multipath zone. While antennas configured in this manner will cost more than the "off the shelf" variety, that is money well spent. The reader is urged to explore this matter in more detail with antenna manufacturers when the next opportunity to purchase an FM antenna arises.

References & Notes

1. The author is also an engineer with the United States Information Agency (USIA). This paper was prepared independently of the author's Government employment. The material and opinions expressed herein do *not* reflect any official views of the USIA or the United States Government.

2. P.O. Box 277, Fairfax, VA 22030, USA

3. e.g., Stutzman and Thiele, "Antenna Theory and Design", p. 129; published by John Wiley & Sons, Inc., New York, 1991; and Jordan and Balmain, "Electromagnetic Waves and Radiating Systems", p. 427; published by Prentice-Hall, Englewood Cliffs, New Jersey, 1968

4. This subject was explored in considerable detail in the author's paper "Optimization of VHF Effective Radiated Power and Antenna Height Combinations", presented at NAB's 1992 Broadcast Engineering Conference. Copies of that paper are available from the author.

5. After Stutzman and Thiele, *id.*, p. 125, Eq. (3-33)

6. The author thanks John Marino of the NAB Science & Technology Department for suggesting the exploration of this topic.

Table IV

Distance to First Null/Minimum

Antenna Height (ft)	Number of Bays: 2 (mi)	3 (mi)	4 (mi)	5 (mi)	6 (mi)	7 (mi)	8 (mi)	10 (mi)	12 (mi)
100	0.0	0.1	0.1	0.1	0.1	0.1	0.1	0.2	0.2
200	0.1	0.1	0.1	0.2	0.2	0.3	0.3	0.4	0.5
300	0.1	0.2	0.2	0.3	0.3	0.4	0.4	0.6	0.7
500	0.2	0.3	0.4	0.5	0.6	0.7	0.7	0.9	1.1
700	0.2	0.4	0.5	0.7	0.8	0.9	1.0	1.3	1.6
1000	0.3	0.5	0.7	0.9	1.1	1.3	1.5	1.9	2.3
1300	0.4	0.7	1.0	1.2	1.5	1.7	1.9	2.5	2.9
1800	0.6	1.0	1.3	1.7	2.0	2.4	2.7	3.4	4.1
Depression Angle (deg)	30	19.5	14.5	11.5	9.6	8.2	7.2	5.7	4.8

Table V

First Null Proportion of Potential Service Area

Antenna Height (ft)	Number of Bays: 2 (%)	3 (%)	4 (%)	5 (%)	6 (%)	7 (%)	8 (%)	10 (%)	12 (%)
100	0.00	0.00	0.00	0.00	0.00	0.01	0.01	0.01	0.02
200	0.00	0.00	0.01	0.01	0.01	0.02	0.02	0.04	0.05
300	0.00	0.00	0.01	0.01	0.02	0.03	0.03	0.05	0.08
500	0.00	0.01	0.01	0.02	0.03	0.04	0.06	0.09	0.13
700	0.00	0.01	0.02	0.04	0.05	0.07	0.09	0.15	0.21
1000	0.01	0.01	0.03	0.04	0.06	0.09	0.11	0.18	0.25
1300	0.01	0.02	0.04	0.06	0.09	0.12	0.16	0.25	0.36
1800	0.01	0.02	0.04	0.07	0.10	0.14	0.18	0.29	0.41

GRAPHICS AND EDITING

Sunday, April 18, 1993

Moderator:
Bob Ogren, LIN Television Corporation, Providence, Rhode Island

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Chuck Dages
CBS Television
New York, New York

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Lee Spieckerman
LIN Television Corporation
Fort Worth, Texas

*Paper not available at the time of publication.

DISK TECHNOLOGY APPLIED TO THE EDITING ENVIRONMENT

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ABSTRACT

The use of hard disk drives in TV production has been growing since the early 1980s. Aside from their early applications for still storage they are now increasingly used as video recorders. The applications vary between the linear and true random access recorders the latter being an integral part of concurrent editing systems. The very latest technology offers longer storage in a smaller package and underscores the bright future for disks in the editing environment.

DISKS, AN OVERVIEW

Disk drives hold a special position in the post production environment. Reliable, high capacity disk drives are a necessary part of many of the systems now used both in production and post. For example still stores and graphics systems both depend on disks for quick access storage of hundreds or thousands of pictures. Without them their development would have been delayed or even not happened at all.

Today a whole post production industry has grown up dependent on the use of disks. Broadcast quality graphics can be created and stored within a network of devices all talking via disks. 3D images, rendered frame by frame, can be stored onto disks for real-time play-back. Edits can be planned in a low resolution disk based off-line suite, right up to on-line editing of broadcast quality images performed in a random access disk based environment. In each case disks provide the most cost effective solution for the required working storage.

ORIGINS OF DISK BASED SYSTEMS

In 1980 the first practical still stores became available based around SMD interface Winchester disk drives. The state of the art in disk storage then came in the form of 14 inch platters sealed in a 6 RU, 19 inch rack mount unit (power supply extra) capable of holding 80 MBytes of information. (Using the 4:2:2 component digital coding system the number of pictures stored can be estimated as slightly more than one per MByte.) Shortly after the first still stores, paint systems appeared using the same disks as caches for work in progress. To begin with capacities increased only slowly, from 80 to 160 MBytes and on to 330 MBytes. Next the platter size shrunk to 10 inches while the capacities expanded to 475 MBytes and, by the mid 1980s, on to 700 MBytes. The data transfer rates of the faster disks allowed picture recalls to be completed in around half a second. Both capacity and speed were ample for stills but for application as a video recorder the pictures would have to be recalled at video rate.

If disks could be used in place of video tape recorders there were important advantages that could be expected. Beside offering direct access to the data, the very few moving parts - two in all - would contribute to give a very high degree of reliability. The heads fly above the disk surface, not making contact, so there is no wear of the medium. However many times it is written or read the integrity of the original data is retained. There would be no dropouts.

There would be disadvantages. The sizes of disks tended to be quite large, especially when measured against other recording media such as video tape. Despite their excellent MTBF disks still had to be handled with a degree of care. Excessive vibrations could send the flying heads crashing into the disk surface.

Besides these points it has already been mentioned that there was finite capacity and data transfer speeds did not match up to the needs of video. It would take a revolution in disk technology to advance the use of disks and so bring their benefits a wider video market.

GRAPHICS COMPOSITING

In the mid 1980s parallel transfer disks became available, offering access speeds at the level required to make the storage and retrieval of video frames possible at video rate. Capacities still remained relatively low, limiting the practical applications to areas where only short video clips would be used, but the stage was set for the video disk revolution. Storage capacities of 30 seconds were normal, with some dedicated systems offering as much as 2½ minutes. Attitudes to the use of disks varied between equipment manufacturers as well as users. This difference represents the point of divergence in the application of disks for editing.

Those with their eye on the traditional system design realised that replacing digital VTRs with digital disk recorders (DDRs) could be a more cost effective way to provide digital sources. At the same time they effectively removed the VTR's mechanical overheads to offer advantages of zero pre-roll time, immediate frame access and no dropouts. But system operation was still restricted in that it remained a linear, video rate, process.

For others the new disks had more to offer. They gave the opportunity to make a more fundamental change to ways of working, breaking away from the limitations of the operating procedures dictated by tape. From this idea a new operational environment developed, digital compositing, and a machine called Harry™. Rather than attempting high density graphics compositing work within a traditional edit suite, it could be undertaken in a system designed and dedicated to that specific task. The benefits were in the speed and style of operation, the unique range of facilities that could be offered and a lower cost than that of a conventional suite. By considering disks as a component fully integrated within the equipment, much as an integrated circuit is a part of electronics, it was possible to create a new operating environment, rather than simply an alternative form of recorder.

Harry™ treated its disk storage in a very different way to the traditional suite where each video source and destination is supplied by a separate machine be it VTR, DDR or some direct input. The Harry™ disk system was specially designed to provide video rate random access for sources and destinations, from the one store. So, for example, a cut edit did not require the copying of two source clips but merely the replay of the original material in a different frame order. This saved both time and disk space. At the same time disk management was much simplified as there was no need to move material to the right disk, or even the right part of a disk, to achieve the required combination of sources to be available for a task. The more complex the task and the more sources needed, the greater the advantages of random access. As there was no constraint introduced by fragmentation - the disk capacity can always be used to its full extent. A single clip could be scattered anywhere over the disk and then used without restriction.

A look at the design of edit suites using DDRs shows, in general, similar thinking to that applied to analog suites. Their block diagrams and operating procedures are similar, if not the same, as those in the common, traditional model. Looking at the digital compositor, the new environment has a very different layout and method of operation which is directed at servicing a more specific range of production requirement than the traditional suite.

CONCURRENT EDITING

The disk handling techniques developed in Harry™ enabled a re-think of the conditions required to create an edit suite. By combining the ideas for random access video employed for graphics compositing with the multilayer capabilities of traditional edit suites a new editing philosophy was created, concurrent editing.

Although this paper deals with the technology of disks, disks alone do not make an edit system. When Flash Harry™ gained five minutes capacity editing became possible within it - but it was still regarded as a graphics compositor. Primarily it was the operational restriction of working one layer at a time that inhibited the interaction required for editing. This was chiefly imposed by the limited data transfer rates of the disks for multilayer operation rather than by their capacity. The goal of an on-line, random access editor remained unachievable until both disk capacities and, most especially, data transfer rates were increased.

This was eventually made possible by a merging of enhanced disks and new developments in their interface to augment their performance. From this combination of parallel transfer disks and the latest RAM and application specific electronics, Quantel developed Chatter™ Technology. This led the way to the world's first concurrent editor, Henry™, providing five minutes of storage for simultaneous multilayering and random access.

RANDOM ACCESS

Concurrent editing is only possible through random access to video. To work simultaneously with multilayers and multiprocesses, without having to duplicate equipment for each layer, requires true random access. As the complexity of work grows the system can cope without needing additional equipment. On the other hand, working with linear disks requires video clips to be cached and copied between different disks as the playing order dictates: a task that takes time proportional to the length of the clip. As the complexity of work, and especially layering, grows a decision has to be made whether to invest in more equipment or to work layer by layer; a method that has already been found not to be ideal.

Another benefit of random access is that single frames or whole clips can be used in any number of layers or processes at the same time, without having to copy or rearrange their storage. In general management is far easier, operation faster and far more flexible in a random access environment.

SCSI DISKS

While parallel transfer disks were dominating the high data transfer rate markets, SCSI disks were replacing SMD disks in areas of data storage where access times were not as critical. Today SCSI drives have been developed to a very high degree. The still image machines, still stores and paint systems, can now store many images on small sized, high capacity disks. For example Quantel's Picturebox stores over 500 pictures on a half height, 3½ inch, 525 MByte drive. This is very compact; only 1½ inches high consumes a mere 12 VA. No longer is it necessary to house the disk system in a separate box as it can easily be mounted internally. At the same time full height 8 inch external drives have moved up to 2 and 3 GByte capacities to provide useful bulk image stores.

While SCSI disks alone would not provide the necessary data transfer rates for video rate access - the 525 MByte model mentioned still takes around a third of a second to transfer a picture - connecting a number of drives in parallel could, collectively, mimic the operation of parallel transfer disks. This technique has been used by a number of manufacturers to produce the next generation video rate disks. Such an array of disks can offer greater capacities both for DDRs and, through the application of special interface technology, for random access stores.

CLOSE COUPLED CHATTER TECHNOLOGY DISKS (DYLAN)

Quantel's approach to SCSI based disk recorders has benefited from the experience gained with parallel transfer disks and chatter interface technology. The combination of an advanced SCSI disk array with the electronics of chatter disk management, named Dylan, has improved the benefits available. True random access is the main advantage, combined with a data transfer rate in excess of video, as well as 15 minutes of storage. Such a disk system is contained within two identical packs each only of 5 RU height in a 19 inch rack frame. The total power requirement is a nominal 800 VA, which compares very well with 4 kVA for the five minutes provided by the parallel drives. Clearly the SCSI disk array as described already has very significant advantages.

Physically each Dylan disk pack comprises a single frame divided into two. The lower section contains 20 standard 525 MByte SCSI drives connected via short SCSI cables to the chatter technology cards housed in the upper compartment. By employing standard SCSI disks use can be made of the diagnostic and error correction capabilities included in the drives, resulting in error free communication. Redundancy can be built into the disk pack, allowing the failure of a complete drive without affecting the working of the system. Not only is this of obvious operational benefit but it also allows servicing to be fitted around the work, not the other way round, as is too often the case. When the offending drive has been replaced the missing data is reconstituted using the same method as employed for error correction.

CONCLUSIONS

Disks have been in use in video equipment, in and around edit suites, since the early 1980s; the application then was solely for the storage of stills. That use has flourished, grown and benefited from the later developments in disk capacity and reductions in size and power. At the same time the stills and graphics systems themselves have become far more widely used.

Within the editing environment itself, disks can occupy two distinctly different places. In one they can be used as a straight alternative for VTRs, offering the benefits of instant access and zero pre-roll times but, essentially, with their linear operation, no change in functionality. In the other they can provide true random access and become a fully integrated component within an edit system.

The random access close-coupled chatter disk system provides multiple sources and destinations from the one disk array at above video rate. Edits can be performed without the need to use multiple devices. There is no need of external disk management, other than recording in of source material and playing out of results. Disk space is used in the most efficient way possible with no need to move clips to the right place or any restrictions due to fragmentation. As each clip can contain non-contiguous frames the available capacity is effectively increased.

It was through the random access disk systems and their integration that it became possible to develop concurrent editing. The physical attributes are of secondary importance but the introduction of the SCSI disk array in Dylan has significantly cut size and power consumption as well as greatly boosting the storage time. The concept of concurrent editing is now only one year old but its future is already looking assured, especially through these later developments of disks, interfaces and integration.

A REAL TIME, DATA DEPENDENT, HYPERMEDIA DISPLAY SYSTEM FOR ELECTION GRAPHICS

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CBS has, for many years, been using computer control of television graphics equipment to display results on election nights. These displays are based on real time data. Media Computing produces PC-based products for the collection of this data and the control of such graphics. At the 1987 NAB Conference, CBS presented a paper outlining the history of election graphics there and the control system then in use. The growth of multimedia systems made us realize that, in fact, what we had produced was a hypermedia display system based on real time data. In 1991, CBS hired Media Computing to produce a state-of-the-art display control system based on networked PC's.

This paper will consist of two parts. The first will describe the creation of the on-air displays and the meaning of the terms in the paper title. We will also describe the design process leading to the current control system. The second part will describe the current election graphics system in detail, including how distributive processing is implemented on a LAN, hardware redundancy, multimedia control and software techniques.

ON-AIR DISPLAYS

Election night broadcasts, especially in presidential election years, require broadcasters to convey a great deal of information to their viewers. This is even more important at the network level where we are called upon to inform the nation as they choose their national leaders. This information must be presented so it is clear, unambiguous and visually interesting. Conflicting with this is the requirement that displays be available quickly.

CBS News has been using computers for election graphics displays for many years.[1] The technology used in these displays has always been on the cutting edge of broadcast television graphics equipment available at that time. The following table outlines some of these advances and the dates used by CBS.

Date	Graphics System
1970	Modified IBM terminal with text information shot with a live camera.
Mid 1970's	Electronic character generators control room keyed over live camera background.
1978	Electronic still store replaces live camera. Both foreground (data) and background are computer controlled.
1980	Ampex Video Artist (AVA) paint system used manually.
1982	Dubner CBG-2's used for animated displays. Both data and background were generated in one device for most displays.
1986	Abekas A-62's used for animated backgrounds. Chyron 4100's displayed the data. Full resolution was now available in the background graphic. Stand-alone keyers presented a complete display to the Control Room (see Figure 1).
1988	Four graphics devices plus three keyers were used per Display System. This combination created displays with animation in all elements.
1990	Chyron Infinit was used for the data. Animation was now simpler in the data as well as the background. This was also the first election with data supplied by Voter Research and Surveys (VRS).

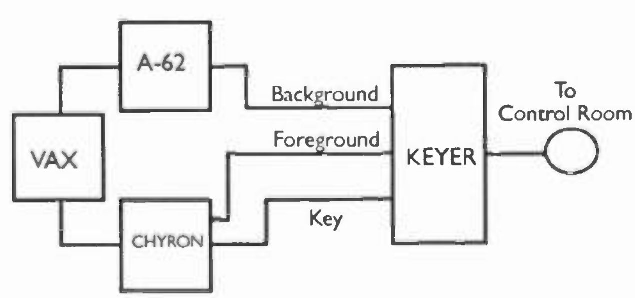

Figure 1. 1986 System Block Diagram

HYPERMEDIA?

Hypermedia is defined as:

> *A computer-based information retrieval system that enables a user to gain or provide access to texts, audio and video recordings, photographs, and computer graphics related to a particular subject.*[2]

The CBS Data Acquisition and Graphics Control System has for many years been such a system. The system has permitted the broadcast production staff (i.e., producers and directors) to request any information display, in any order, with minimal preparation. The subject in this case is a particular race on Election Night or some other political broadcast. The displays are made up of text (names and numbers), video recordings (backgrounds recorded in the Abekas A-62's), photographs (candidate pictures) and computer graphics (as rendered in the television graphics devices) (see Figure 2).

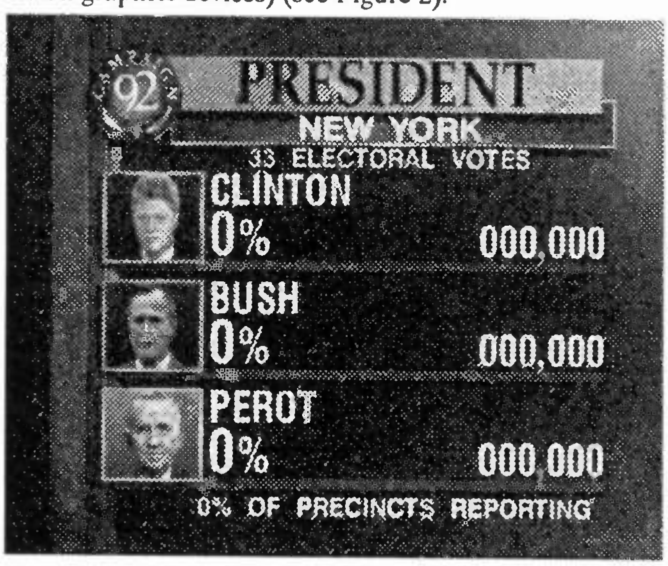

Figure 2. Typical 1992 Election Raw Vote Composite Display

These systems have been as real time as the technology of the time allowed. The Control System advanced from PDP-11's in the early 1970's on to VAX super minicomputers. The custom applications programs were developed and added to as the on-air displays became more complex. More processor time was required to create and control these displays. Data base servicing took on lower priorities. The result was that, although displays could be more complex, the data processing was now less efficient.

As an example, many of the complex animated displays generated in the Dubner required several minutes of preparation. Changes to the order of the display was not generally possible. The Abekas provided a way of randomly playing background sequences with significantly less preparation time. The unit needed to be prepared only to play the next sequence. This permitted late changes to the order of display with a slight time penalty. The Chyron 4100 character generators were simply typing text with the computer acting as a fast typist.
As we began to use the Infinit, taking advantage of its animation capabilities, it also required some preparation time. The combination of the Abekas and the Infinit provided visually interesting displays with less preparation than the Dubner alone. We were approaching the goal of real time displays.

All election broadcast on-air displays are presenting data from several sources (see Figure 3). Raw vote information has, for a number of years, come from the News Election Service (NES). Actual votes counted are gathered by NES and then transmitted to the broadcasters and others. Analyzed vote information such as prediction of winners and exit polling is provided by Voter Research and Surveys (VRS). This organization was formed prior to the 1990 election to consolidate this process for all broadcast networks. Previously, the networks each did their own analysis and exit polling. Their information comes from surveys taken at selected polling places. This data is combined with raw vote information and analyzed in a main frame computer system. Both organizations broadcast this data on 9600 bps (bits per second) dedicated data circuits.

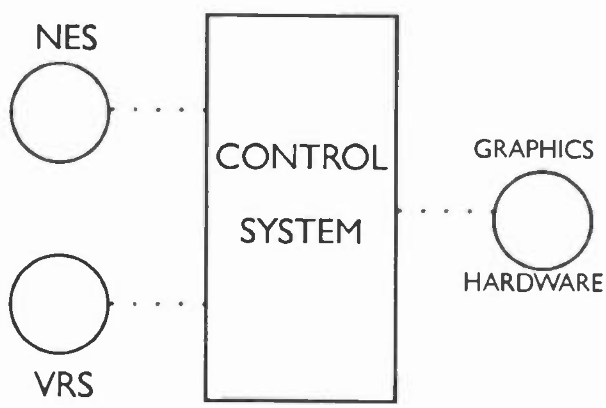

Figure 3. Data Flow Diagram

The CBS Graphics Control System will make some display decisions based on the data. The system will, for example, decide on order of candidates in a display depending on the current vote percentage.

The system is, therefore, a Hypermedia system that is data dependent and real time.

DESIGN PROCESS

After the 1990 election, CBS News determined that it was desirable to move to a simpler on-air graphics "look" for the 1992 political year. In addition, the Data Acquisition and Graphics Control System then in use required modification.

The entire system existed on a pair of VAX 11-750 super minicomputers running under VMS. The applications were written by a group of programmers employed by CBS News. All processes ran in a multi-tasking environment that required attendance during a broadcast to insure that task priorities were shifted as required. As the on-air displays and transitions became more complex, this environment became overloaded. The result was that some tasks received less attention than they required. Control of the on-air displays was given priority over data acquisition and processing. Since CBS News as well as the other networks were each doing their own predictions, the information could easily be manually groomed in the Graphics System.

As the networks joined together to form VRS, they received the prediction information simultaneously. The goal was then to process this data quickly to get the information to the public. Unfortunately, the VAX-based system could not process the data and control the graphics devices in a timely manner. The programming staff that had produced this system was now no longer employed by CBS News.

In early 1991, a small group consisting of CBS Engineering and CBS News personnel began outlining the requirements for an ideal system. A Design Target Specification was produced and sent to 15 potential vendors in late summer 1991.

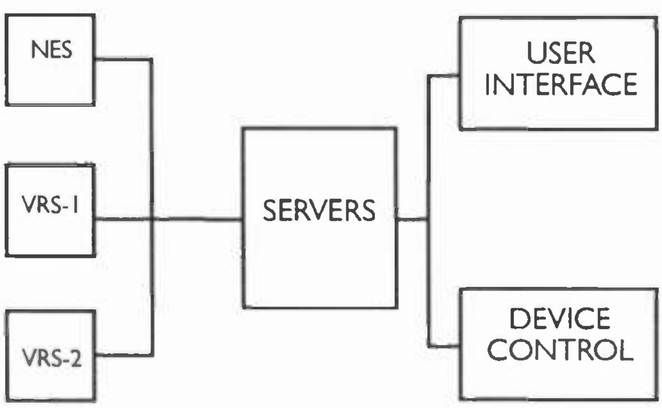

Figure 4. 1992 Control System Proposal

The goals were ambitious. We were specifying a system of networked personal computers (PC's) that would gather the data, analyze it as required and control the graphics devices (see Figure 4). We wanted to purchase hardware that could be reused between elections. It was desired to use standard PC and network operating systems as well as a user programmable graphics control language. There was also a desire to have a system automate some of the formerly manual processes in the on-air Control Room.

In November 1991, CBS News chose Media Computing of Phoenix Arizona to supply the system. The computer hardware was purchased from Northgate. The network is a 10Base-T variant from LANNet. The schedule had the first broadcast for the New Hampshire primary on February 18, 1992. In fact, we used the system first during the America On-The-Line program following the State of the Union Address on January 28, 1992.

CURRENT ELECTION GRAPHICS SYSTEM

Hardware

Local Area Network-Based System As mentioned above, the CBS Election Graphics System is based on PC distributed processing. In this type of environment, each particular task or function is performed by a particular PC. When you have lots of tasks or functions to be performed simultaneously, you distribute them among lots of PC's - in CBS's case, 50 PC's! The distribution is performed over a local area network (LAN). Each PC has a network interface card installed in one of its expansion slots. Each of these cards is attached to the network cabling system which, with the help of the LAN software and LAN file servers, enables the PC's to communicate among themselves. The file servers run the network operating system to regulate communications among the PC's attached to it and allow multiple PC's to access the same files and/or data bases at the same time.

Server Redundancy Due to the critical nature of the system, it is designed with absolute total redundancy. And in the case of super critical components, a primary and three real time backups are employed. For example: the file server uses a technique known as disk duplexing - copying data onto two hard disks simultaneously, each via a separate disk controller and interface cable (see Figure 5). If any of these components fails, the other disk continues to operate without data loss or interruption. A side benefit of duplexing is the ability to split multiple reads between the disks for fast simultaneous processing.

But what happens if the entire file server becomes inoperable? The system design addressed that possibility by employing dual, identically configured, file servers in real time. Each time a PC on the LAN is turned on, it loads a TSR (terminate and stay resident) program which intercepts all of the PCs requests for file server disk access and processes them in a dual file server environment. In the case of writes, the TSR sends the data to the primary server then it sends an identical copy to the secondary server. In the case of reads, only the primary server is accessed. If either of the servers should fail to respond, the TSR switches over to the remaining operational server. This is all done in a manner completely transparent to the user and/or the application running on the PC.

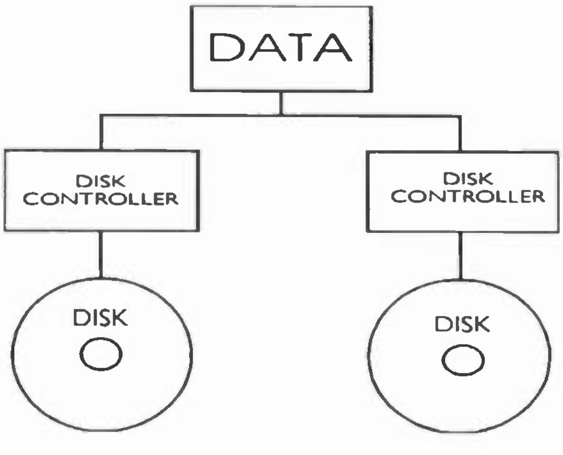

Figure 5. File Server Disk Duplexing

<u>Work Station Redundancy</u> In addition to server redundancy, the system also uses two types of PC work station redundancy. The first is a dual path between each of the work stations and the servers. The network interface card in each work station (and in each server) is attached to an external network transceiver - one transceiver for each network card (see Figure 6). The transceiver is also attached to a primary path cable and a secondary path cable. If the primary path should become inoperable, the transceiver automatically (and transparently) changes over to the secondary path.

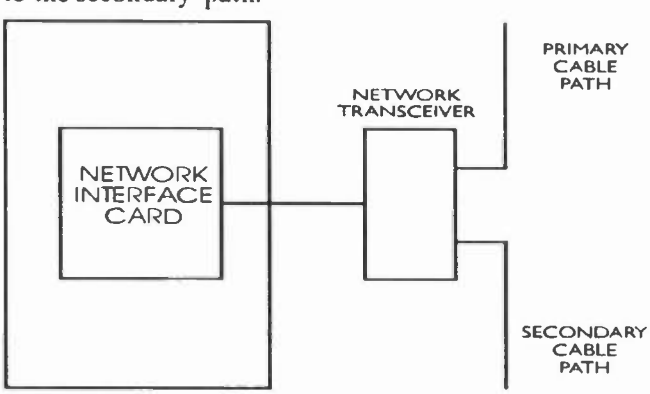

Figure 6. Work Station' Dual Path

The second type of work station redundancy is interchange-ability. With the exception of the work stations attached to the digital disk recorders, any work station can perform any function from any place on the network. The work stations "personality" and NetBios ID are determined AFTER a function is selected from one of the menus, not when it logs onto the network. Graphic control, for example, was performed from the third floor R&D suite during development; from the first floor training room during testing; from Control Room 47 on election night; from Control Room 43 on the second floor the morning after the primary elections and once from the wire input room during a multiple hardware failure emergency. (The failure occurred before all of the redundant hardware was installed.)

Of the 50 PC's attached to the network, only 12 of them have human operators - the rest of the PC's perform automated functions. Nine of these operators (three technicians controlling system output, four producers and two system managers) use a 128-key programmable keyboard to help them navigate through the 200 plus system functions. By using a concept similar to having shift and control keys, one key is able to perform multiple functions. For example, pressing the **Arizona** key by itself tells the system which state the next display will come from. Pressing **Shift** and **Arizona** tells the system to display (on the operator's PC) the current status of all races in Arizona. Using this technique with color coded key caps, macros typing up to 30 key strokes and a few days of practice produced very fast and accurate operators. The system design is such that all 10 keyboards (including the backup keyboard) are programmed identically even though they appear to handle drastically different functions.

<u>Multimedia Control - Graphic Output Subsystem</u>

<u>Output Process</u> All of the election displays are generated by four Graphic Output subsystems. One of these systems is dedicated to producing maps of the U.S. showing a compilation of results of all the state races for president and electoral vote count.

The other three subsystems (see Figure 7) generate any of the 2400 plus displays with the latest data. Each of these displays (see Figure 2) is made up of the election data in the foreground produced by the character generator, and an animated background produced by the digital disk recorder.

Display lists (see **List Management** below) are created by any and all of the nine operators with the programmable keyboards. As the lists are being created, **ANGIS** (the election/news graphics data base) is automatically checked to make sure the list entries are valid. These lists are

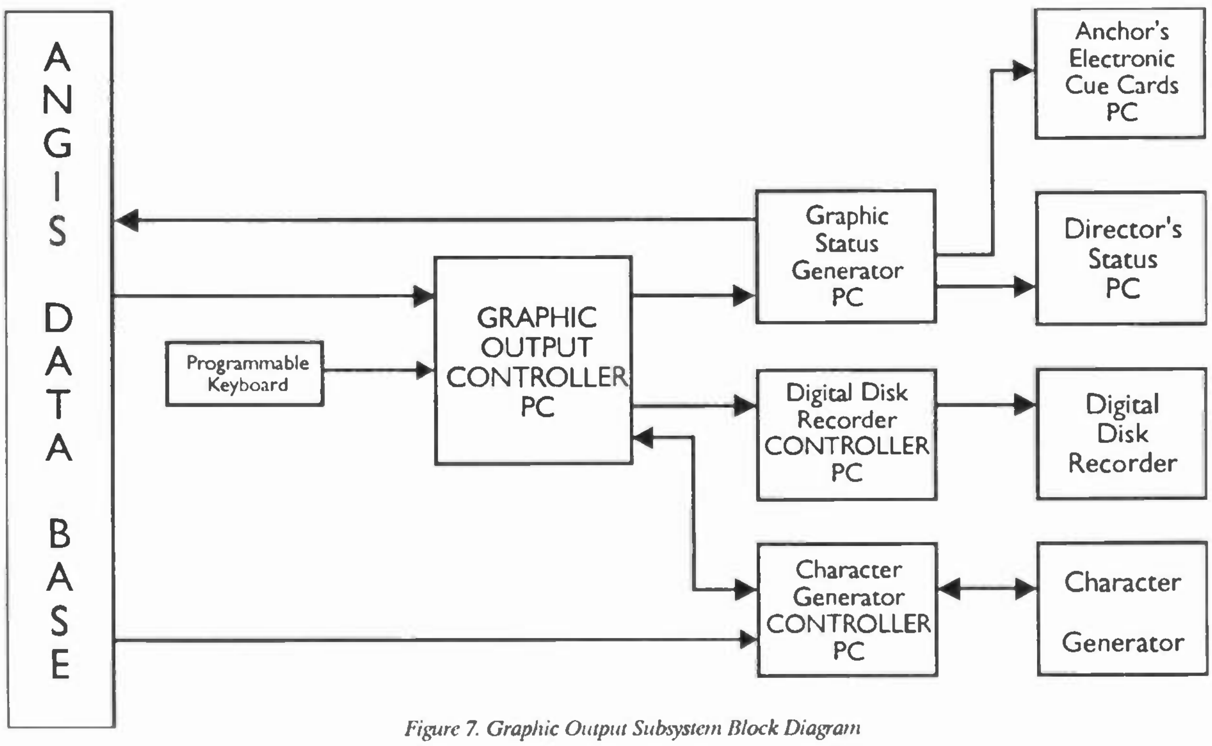

Figure 7. Graphic Output Subsystem Block Diagram

processed by the Graphic Output Controller PC which coordinates all the PC's involved in producing the displays and related status. As each entry in the list is processed, a short code is sent to the Character Generator Controller PC telling it which race is to be aired, what type of display is to be used (raw vote, estimate or winner), and what type of transition is to be used to bring the display to air. The Character Generator Controller PC gets the appropriate information from the **ANGIS** data base, sorts it so the winner is always on the top of the display, translates the data into a language the character generator can understand and sends it out its RS-232 port to the character generator. At the same time all this is going on, the Graphic Output Controller PC also sends a code to the Digital Disk Recorder Controller PC telling it which animation is to be aired. This PC uses a lookup table in RAM to find the appropriate command to send to the digital disk recorder via a microprocessor- based I/O card.

Status Process When the character generator starts its transition, it sends a message to the Graphic Output Controller PC (via the Character Generator Controller PC) triggering a status update. The Graphic Status Generator PC processes the status information for three different types of status displays.

The first is sent to the Director's Status PC. This PC displays, in very large characters, an indication of which race is on air and the next two races to be aired. The second type of status is entered into the **ANGIS** data base. This allows the producers and system managers to see this status information on their PC's screen. The third type of status is sent to the Anchor's Electronic Cue Card PC. This PC uses the status information as a pointer into a cue card data base. As the viewers at home see a new display, the anchor sees an electronic cue card of background information on the candidates and race that is on air. The PC also shows the cue card for the next race to be aired so the anchor doesn't have any surprises.

From the beginning of the output process (the time a display is requested) to the end of the status process is only three and a half seconds!

Input Processing Subsystem

The input to the Election Graphics System is totally automated. All of the data enters the system from two sources: News Election Service (NES), which supplies raw vote information, and Voter Research and Surveys (VRS), which supplies winner predictions and exit polling. Both services transmit their data at approximately 960 characters per second in a very condensed format (see Figure 8).

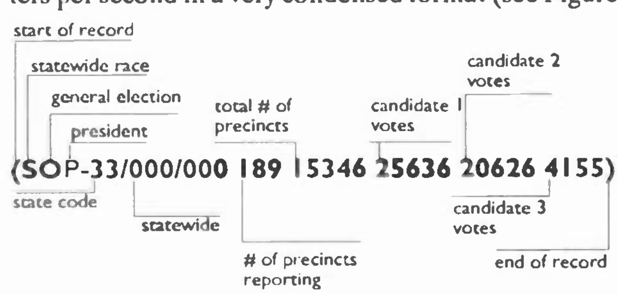

Figure 8. Example of NES Data

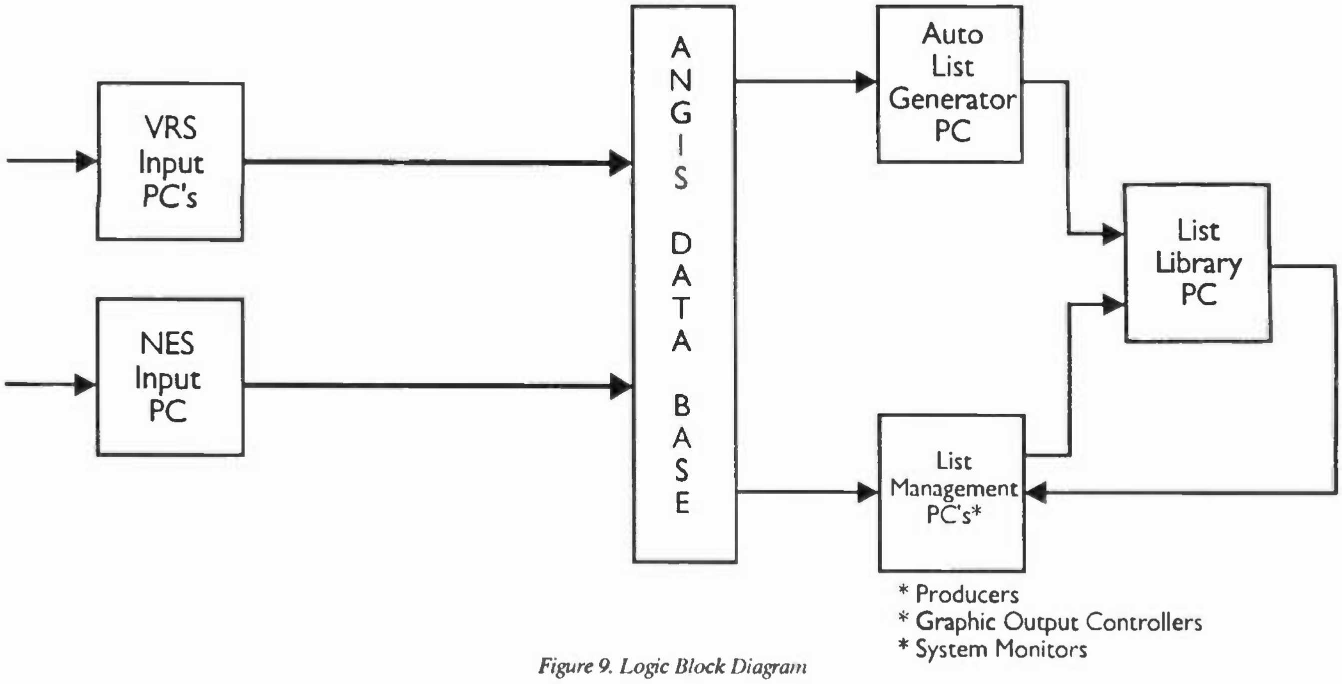

Figure 9. Logic Block Diagram

The NES Input PC (see Figure 9) searches through the massive amount of data coming in and extracts those records CBS wishes to process. When the NES Input PC detects a desired record, it calculates the percentage of precincts reporting, the percentage of vote for each of the candidates and writes the updated information into the appropriate **ANGIS** data base record which already contains the race name, candidates names, etc.

VRS sends its information over two connections which are processed by two VRS Input PC's. When the VRS PC's detect desired information, they determine: 1) if a winner has been declared and if there is a winner, who that winner is and if it is OK to release the information; 2) if a winner is about to be declared; 3) if a candidate is leading, who that candidate is; or 4) the race is too close to call. Once these determinations are made, the data base is updated appropriately.

List Management

The input process discussed above provides the system with the raw data. The next step is organizing all this information. This organization is done through list management. The lists themselves are very simple. Each entry consists of state name, race type (President, Senate, etc.), type of display (raw vote, winner or estimate) and its location in the **ANGIS** data base. The lists are generated by two very different techniques. The first is an auto list generator, and the second is the more conventional way, via the human interface.

Auto List and Map Generators The system is designed to automate as many tasks or functions as possible, leaving those tasks to humans that only humans can perform. The Auto List Generators produce some of the required lists much faster (and much more accurately) than humans. These automated lists include a list of all the presidential races where winners have been declared. The Auto List Generator PC constantly searches the data base for winners. When it finds one, it adds it to the list. It keeps doing this until the list is aired, once the list is aired, it starts a new list. This same technique is used for the Senate, Governor and House races.

Another type of automated list is the Poll Closing List. Each time a state (or states) polls close, the Poll Closing PC builds a list of all the declared winners at the time of the poll closing. The system is programmed in such a manner that if a race is included in the poll closing list, it is NOT included in the winner lists described above.
The Poll Closing PC also automatically builds maps. These maps include: 1) poll closing maps - graphically shows the projected presidential race winners for each set of poll closings, 2) Senate map - graphically shows the ongoing status of all the Senate races - as each race is declared, the appropriate state is shaded in the winning party's color and party totals are updated, 3) Governor map - same technique as used in the Senate map, but applied to the Governor races.

The Human Interface As mentioned above, there are nine PC's with programmable keyboards. These PC's are the ones used with the human interface. All nine

PC's are used to produce lists - simultaneously. The programmable keyboard is an integral part of the human interface in general and is very helpful in performing list management functions. Each part of a list entry is made by pressing just one key, e.g., to enter **North Carolina**, the users need only press the **North Carolina** key - they do not have to type the 14 characters. As each of the list entries are made, the system automatically verifies the entry and adds the **ANGIS** data base record location to the entry.

All the expected list functions are available: insert, delete, un-delete, move, swap, cursor positioning and preview. Once a list is produced, it is available to any and all of the nine human interfaces. For example, a list is created by one of the producers, reviewed and possibly edited by one of the system managers then the list is processed on one (or more) of the Graphic Output Controller PC's.

The Graphic Output Controller operator triggers each display in the list on command from the director. The operator also has the ability to update the information that is on air and/or the next display to be aired (the next item in the list). If needed, the operators can suspend a list and air a display which is not in the list, then continue the list at the point of suspension. The operators can air other types of displays, various maps and transformations, by pressing specific keys on the programmable keyboard. For example, if a U.S. map showing the current status of all the Senate races is to be aired, the operator presses the appropriate key and the Graphic Output Controller PC (see Figure 7) sends a message to the Character Generator Controller PC telling it which map file to process. The map files contain hundreds of commands telling the character generator how to color the map and commands telling the Controller PC how to retrieve the latest data. All of this data is processed by the Character Generator Controller PC and sent on to the character generator.

<u>Auto Library</u> With all these lists being generated automatically by PC's and manually by nine people, you might ask how do you keep track of what's going on? Good question. The answer - automate the task with the Auto Library PC. This PC monitors the lists as they are being created (or deleted), puts the lists into groups, sorts the groups and the lists within the groups then creates as many library windows as needed to hold all the list information. All of the PC's running the human interface sense the creation of a new set of library windows and load the first window into the bottom third of the monitors screen. The operators page through the windows looking for the desired list. When the correct window is found, they highlight the desired list and then press the appropriate key on the keyboard to view, edit, air or delete the list.

<u>System Monitoring</u>

Two types of system monitoring are performed - local area network monitoring and input monitoring. LAN monitoring is used to detect faults, critical events, network traffic and statistics. If anything out of the ordinary occurs on the network, visual and audible alarms are activated giving us the opportunity to investigate and resolve the problem before it impacts the airing of displays.

Input monitoring is a lot more subjective. The person who programmed the **ANGIS** data base constantly searches through the records checking the data to make sure it is logical and valid. If an inconsistency is found, the source (NES or VRS) is contacted and a resolution to the problem is discussed then implemented.

<u>How It Is All Accomplished</u>

<u>Hardware</u> All of the PC's on the network are 386 or 486 based. There is nothing special or custom about the PC's or the expansion cards in them - even the intelligent I/O cards in the Digital Disk Controller PC's are readily available through many mail order firms. When the PC's are not in use on election night, they are used by CBS staff in normal day to day operations. The character generators and digital disk recorders were not modified in any manner - the computer interfaces were already included by the manufacturers.

<u>Software</u> The local area network operating system is NOVELL's NetWare version 3.11 running on each of the two servers. The disk operating software running on each of the workstations is Microsoft's MS-DOS version 5.0. The PC's for the producers and network mangers also run Quarterdeck's DESQview version 2.41. This gives them instant access, via the programmable keyboards, to any one of the 47 programs (that is not a typo - **47 programs**) running on the PC. Using this many programs simultaneously gives the users random access to all of the data base displays and list management functions - they can instantly find the latest information about any race in the data base with the press of a key!

All of the election data base and related functions are processed by Media Computing's **ANGIS**. All of the list management functions and control of the digital disk recorders are performed by Media Computing's **PROtec** (see Figure 10). Both **ANGIS** and **PROtec** are completely user programmable. The system was initially programmed to handle *America On-The-Line*, a real time telephone call in show. It was then reprogrammed to handle the New Hampshire primary 21 days later. It was continually reprogrammed to handle different primaries throughout the year and, of course, the general election in November.

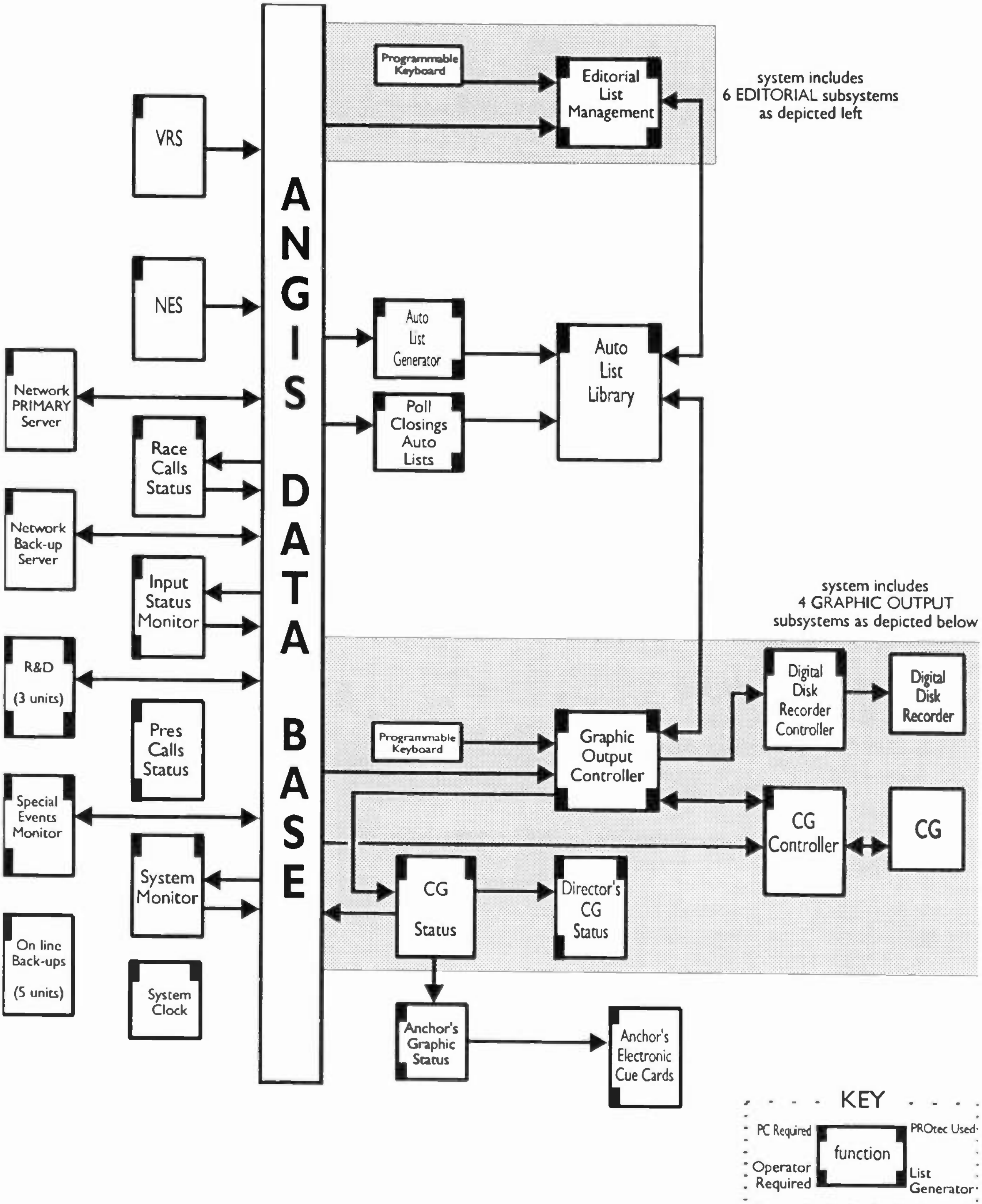

Figure 10. CBS Election Graphics System Diagram

Reprogramming is done through the use of any word processor capable of editing an ASCII text file. Figure 11 is an example of a program segment written in the **PROtec** interpretative programing language.

```
  [SAVE "Ver. date: 11-01-92" IN {DATA 46}]
  [SAVE 112 IN <26>] Display color code 112 = Black on Grey
  .
  .
  .
  [CLEAR SCREEN]
  [PAINT BOX 1, 1, 24, 80, 31] Color code 31 = White on Blue
  [DISPLAY AT 1, 1]
  [DISPLAY {DATA 46} USING COLOR <26>] Version
  [DISPLAY AT 1, 30]
  [DISPLAY " Automated Bumper Map" USING COLOR <26>]
  [DISPLAY AT 3, 10]
  [DISPLAY " Press  B  for Bumper Map       " USING COLOR <26>]
  [DISPLAY AT 5, 10]
  [DISPLAY " Press  S  to  SET time of day " USING COLOR <26>]
  [DISPLAY AT 7, 10]
  [DISPLAY " Press  E  to  End              " USING COLOR <26>]
  [DISPLAY AT 1, 59]
  [DISPLAY "System time:  " USING COLOR <26>]
  [DISPLAY AT 1, 72]
10[COMPUTER TIME SENSE]
  [SAVE COMPUTER TIME IN {TIME CODE 49}]
  [DISPLAY {TIME CODE 49} USING COLOR <26> HH:MM:SS]
  [IF KEYBOARD CHARACTER = "B" THEN 100] Bumper
  [IF KEYBOARD CHARACTER = "S" THEN 090] Set time of day
  [IF KEYBOARD CHARACTER = "E" THEN 920] End
  [GOTO 10]
```

Figure 11. PROtec Program Segment Example

The **PROtec** interpreter reads the program text file into memory then translates the program statements, line by line in real time, into machine instructions the PC can understand and process. Through the use of **PROtec's** menu system (also user programmable) any one of the different programs (***America On-The-Line***, each of the primaries or the general election) can be selected thereby completely changing the systems function at the touch of a key.

By using readily available "off-the-shelf" PC hardware and completely user programmable software, CBS has a system which is easily maintainable and capable of changing to meet new requirements.

REFERENCES

[1] M.A. Harris, "State-of-the-Art Graphics Facility", 1987 NAB Engineering Conference Proceedings.

[2] The American Heritage Dictionary of the English Language, Third Edition.

SPORTS GRAPHICS AND SCORING FOR TELEVISION

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1. ABSTRACT

I think everybody is familiar with pictures showing Sport Results, but not many are aware of the amount of effort required for. Sport Analysis, Graphic Design, Data Base Definition, Selection Screen Method, Today's Match/Competition Initial Data, Match/Competition Data Entry, Screen Selection under TV Producer requirement, etc.

The purpose of this paper is to clarify each of these tasks and to propose different system solutions.

2. SYSTEM REQUIREMENTS

2.1. Real Time Operation:

Sports coverage is live and in many of them the action is so fast that in some situations the system must respond _under a second._. This means no more than a second from when the producer requests a screen to when it is ready.

2.2. User Friendly:

Because of the live situation and the real time operation required the method for calling a screen must be as _easy and intuitive_ as possible. Reducing the operator stress will result in less mistakes during selection and better response times.

We think it should be _sport orientated_.

2.3. Easy to Configure:

The regular CG/Graphic operator should be able to configure the system for today's game within a reasonable period of time.

It would be very convenient to have an interface to _external Data Bases_ so that their data can be used.

2.4. Interfaces to External Devices:

Nowadays most of _the competition data is controlled by some computer_ that interfaces to measurement and timing devices, in which the officials type in data, controls the score, etc.. Considering this _our graphic machine must be connected to that external computer_, so that we will make use of the official data and avoid discrepancies.

2.5. Expendability and Flexibility:

It must be easy to add new functions (ie. new statistics), new graphics (ie. a new team), to do new sports using the same graphic style and most of the graphic elements, to migrate into a new graphic look using most of the job done previously.

3.- IDENTIFIED TASKS

3.1. Sport Analysis

The sport to cover must be studied by an team composed by a sport expert and a TV producer assistant in order _to define the different screens that can be required to give to the TV viewer all the sport information,_ ie.: Line up, Player Biography, Score, Statistics, etc. Of course the screens vary from sport to sport but also each TV producer will stamp his personal style on the game.

Fig 1 reflects the kind of information that this team produces

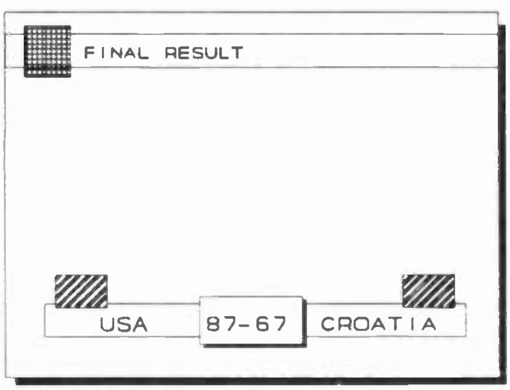


Fig. 1

3.2. Graphic Design

A team of Graphic Artists must design the _Look of Pages_, taking into consideration the previous study, Screen Legibility, Machine Limitations and TV station personality. This means defining: Typefaces, Edges, Colours, Backgrounds, Graphic Logos/-Flags, Layout, etc. The best way of doing this is to simulate some pages in two or more different proposals from which the producer will select one.

For building these proposals it is sufficient to use any Paint System but it is much more efficient to do this in the same machine that will be used in live situations since it is the best way to take into account the possible machine limitations.

An ideal system for doing this job would have the following features:

> HW architecture composed by Independent Graphic & CG Planes combined by a DLK:

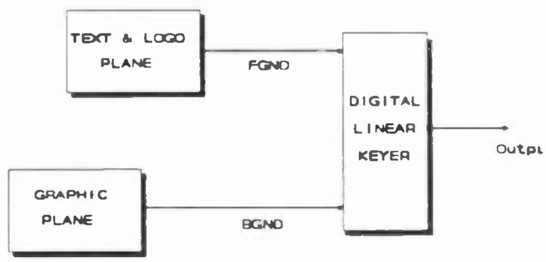


Fig. 2

> Pixel and Object orientated Paint SW.
> Frame Grab.
> On line Font Rendering with horizontal distortion, (for controlling the row distribution).
> Full control over transparency.

Special consideration must be given to the width of the variable fields considering that the number of characters they can house is variable with the content due to the kerning, so that if we define a width for the worst case we may not have enough horizontal space, whilst if the width is not enough there will be uncontrolled truncation.

Once the decision on the graphic style has been made it is necessary to build all the graphics elements: Sport and Teams Logos, Backgrounds and other auxiliary items

Considering that many of today's designs do use some transparency controlled backgrounds for increasing legibility, when inserted into real live video, it is very important to study the possibility of dividing this background in parts that it can be tied together for building different background, for avoiding memory/storage problems. This is possible to do very often due to the vertical redundancy (Rows containing the same type of information).

3.3. Template Construction

Having decided the graphic style and designed the different graphic elements, the graphic operators team must built all the templates defined in the Sport Analysis.

A template is a page composed partially by fixed text/graphics and partially by variable fields:

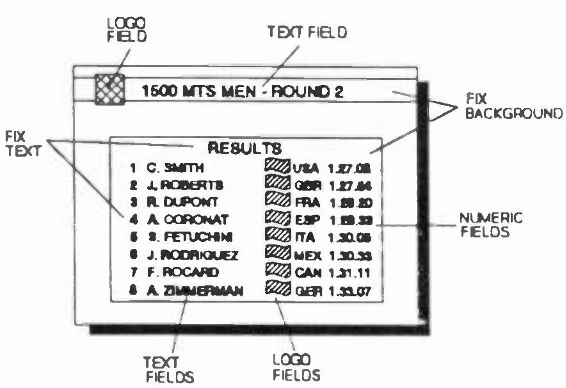


Fig. 3: Template Fields

These variable fields can be of different types:

> Text
> Numeric
> Graphic
> Statistics

Font, colour and justification are predefined, although they can be changed dynamically afterwards. The difference between Text and Numerical Data is in kerning, as the numbers in a numeric field do have a fixed width for achieving good vertical alignment.

The graphic fields do have a reference point to which the reference point of the graphic element will be attached. For vertical alignment purposes it is convenient to build graphic elements of the same group in the same size.

A special consideration must be given to the statistics since this is playing a more and more active role in the sport data that is given to the viewer. A statistics field is defined by its type (bars, pie, etc) and given the data (at the time of the competition) it is automatically built.

Considering the special role of the statistics PESA has added a new feature to it: _contest sensitivity_, this characteristic creates statistic information with objects meaningful to the situation, ie.:

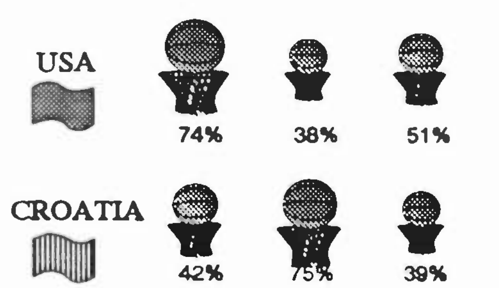


Fig 4: Sport Contest Sensitive Graphics

All the variable fields must be identified so that they can be filled with data coming from a local database or an external computer. This ID can be done by numbers (which is simpler for the computer) and/or labels (that is simpler for the operators).

Three tools are of the utmost importance for increasing the productivity of this phase:

> *Graphic masks*: In a object oriented system it is possible to have a background and independent objects. Because of this by building reference masks it is very easy to position correctly the text plane and the graphic objects.

> *Grid*: The total accuracy is achieved by using a definable grid in collaboration with the graphic masks. The reference points of the object will be attached to the grid points.

> *Row Copy Function*: (because the vertical redundancy), including some automatic renumbering of the field's ID.

> *Print* : For quick checking the field ID and the matching of them with the DB Field.

3.4. Data Base Definition

Considering that some competition/match information can appear in more than one template (ie a player name) and that the design done will be used for more than one competition/match (of course with different information) *it is very convenient to have a Data Base so that there is no redundancy. By simply filling it with different information the system is ready for another competition/match.*

For doing this it is necessary to define the DB structure: number of fields, name of each one and number of values that it can hold,(See Appendix I).

This DB is for supporting the On Line use of the templates, not a general purpose DB. It's purpose is to be loaded with data from today's game with information coming from a general purpose DB, such as DBIV.

3.5. Screen Selection Method

Considering that the sport *coverage is usually live, the number of screens available can be large and the required time response (subsecond in many cases) the screen selection method becomes a major issue.*

One possibility is to call the pages by numbers by relying on the operators memory and some nemotecnic rules, but we think that there are some inconveniences due to operator, stress, number of keystrokes required, large number of pages, etc.

We think that the best approach is to have a *sport system sensitive*, based on selection screens with buttons arranged and configured for each sport. The buttons can be pressed by using a graphic tablet, a light pen or a touch screen. A button or combination of buttons will display a template filled with the linked DB fields. The linking process is done during the definition of the buttons.

It should also be possible to have buttons for data entry in the case of sports with small amounts of data in which the same operator can select screens and enter results of games.

In the Appendix II there is an example of a main selection and data entry screen for Basketball, from which it can be seen that any operator that understand the game a little bit will be able to handle this machine with reasonable speed.

3.6. DB Initial Competition Data Entry

In a competition there is always a DB somewhere that holds at least the name of the participants of the different teams, referees, etc. or we can create one on a PC with standard DB programs.

For our application this means that we must make use of this data so that we do not have the need of typing all the information required for each match, we will select for our DB the information required for today's match. It is not worth considering a general purpose DB running in the graphic machine because of obvious reasons of maintenance.

Only some auxiliary data like venue, date, etc. will be required and that can be entered locally by the graphic operator.

3.7. Competition Data Entry

Considering that the competition is computer controlled our machine must I/F to it and getting the competition variable data by the time it is going to appear on screen.

That is the ideal situation but there are some limitations to consider:

Speed: sometimes the TV requires fastest data entry than the official result system so that a parallel data entry system must be used.
Interfaces: not all the competitions are 100% computer controlled or the computers are ready to interface to them.
Extra data: there is a trend to show a lot of statistics based on data that is not officially entered so that the TV staff must enter it themselves.

Conclusion: We must connect our machine to have as much official information as possible (rapidly for timing and other measurement devices) and having the possibility of entering other competition data.

An important requirement in interfacing to external data entry computers is that if data changes while it is been shown on screen this change must be reflected on it.

The data required by TV will be entered by the same operator that gives the pages to the producer if the amount of data is not too much alternatively it can be entered by another operator and another extra channel or computer if required.

Sport Spotter: For high speed or large amounts of data sports it is very difficult to enter the data and to follow the competition itself. For solving this problem a man called spotter with some sport and TV skills is used to prompt the entry data operator what to enter.

Two *Talkback channels* are always required :

> Among the spotter, the data entry man and the TV graphic operator (Remember that the data entry man and the TV graphic operator can be the same man).

> Between the TV producer (or producer assistant) and the TV graphic operator.

A data monitor must be available to the Spotter in order to check the correct TV data status versus the official score. The TV operator will follow the action through the general TV production monitor set and the data entry guy does not need any kind of sport monitoring mean.

In the short future the data entry user I/F will be done by vox.

3.8. On line Screen Selection

Once the machine is ready to go the *main task of the operator is to give to the producer the screen required within the shortest time possible*. This task must be done by the TV graphic operator and not by any computer guy not integrated in the TV production team. It can be done by typing the right page number that he must remember or by selecting the right screen button or combination of screen buttons.

An example of a *sport sensitive* selection and data entry screen is given in the *Appendix II*. This user I/F method permits the operator to do the data entry and the selection of screens with less mistakes and better time response.

An important consideration is the difficulty of fast selection of a screen button by using a mouse or tablet pen. This can be avoided by concentrating the buttons of shortest response time in the same selection screen and drawing a mask that will be attached to the tablet.

This task will be triggered by vox I/F as well as the data entry in the short future.

4. SYSTEM SOLUTIONS

Many sports like: Tennis, Soccer, Hockey, and even Basketball can be covered with a stand alone machine considering it has the **sport sensitive user I/F**. With this approach only a Spotter and a TV graphic operator is required.

A simple system is represented on Fig. 1 of Appendix III.

In the case of many statistics and/or I/F's to external devices with a system like the one on *Fig. 2 of Appendix III* will be enough. On it a Spotter, a Data Entry man and a TV graphic Operator are required.

A generic system, covering multiple Data Entry points, multiple Selection Screen hubs, I/F's to external devices and On Line general purpose DB is proposed in *Fig. 3 of Appendix IV*.

On it there are the following elements:

Graphic Machines:

On them the TV graphic operator(s) will enter today's competition initial data and select screens for the TV producer:

External Data Entry Computer(s)

The competition results will be entered on as many data entry terminals as required.

I/F's to Official Computers:

If it is possible the official competition data must be used by interfacing to the computer that controls the score, timing, official data entry, etc.

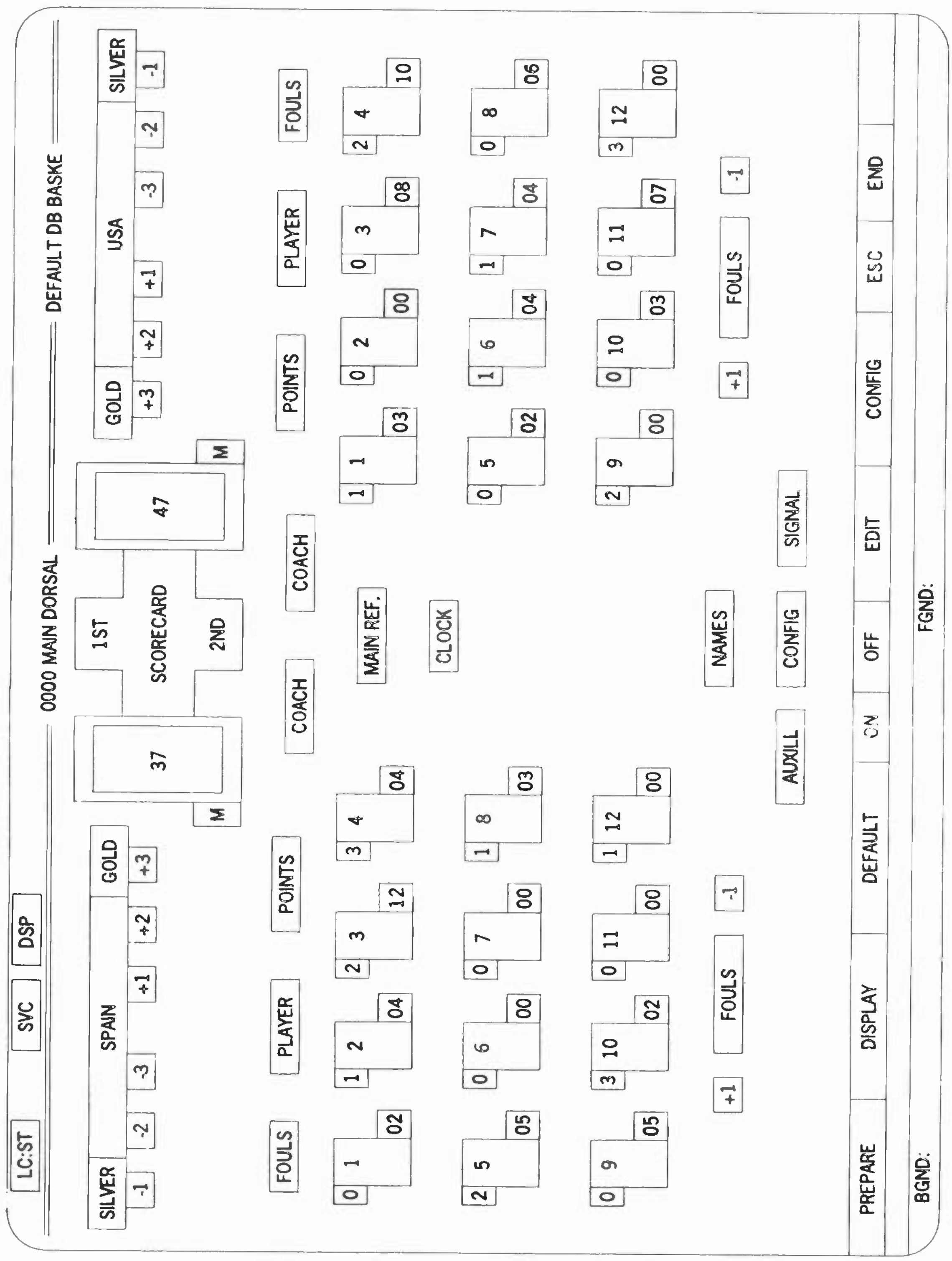
Because the data transfer speed necessary for dealing with tenths of second a special high speed data link is provided direct to the graphic machine, with the goal of not overloading the data network while this running time is active.

APPENDIX I:

TYPICAL BASKETBALL DB

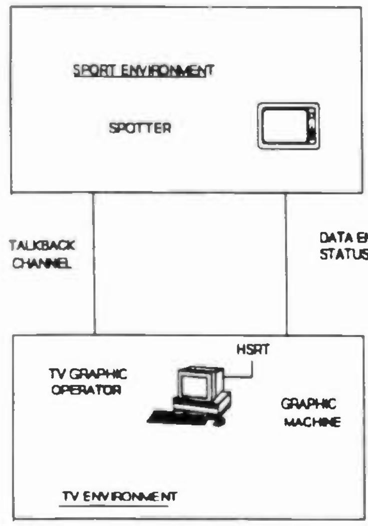
No	Label	Type	Len	Values	
01	TM1_NAME	TX	15	01	
02	TM1_LOGO	LG	01	01	
03	TM1_M_PLYD	NU	02	01	
04	TM1_M_WON	NU	02	01	
05	TM1_M_DRWN	NU	02	01	
06	TM1_M_LOST	NU	02	01	
07	TM1_COACH	TX	15	01	
08	TM1_SCORE	NU	03	01	
09	TM1_FOULS	NU	02	01	
10	TM1_PLYR_NAME	TX	15	12	
11	TM1_PLYR_JNO	NU	02	12	
12	TM1_PLYR_POSN	TX	03	12	
13	TM1_PLYR_AGE	TX	02	12	
14	TM1_PLYR_HT	NU	02	12	
15	TM1_PLYR_WT	NU	03	12	
16	TM1_PLYR_SCR	NU	02	12	
17	TM1_PLYR_FLS	NU	01	12	
18	TM2_NAME	TX	15	01	
19	TM2_LOGO	LG	01	01	
20	TM2_M_PLYD	NU	02	01	
21	TM2_M_WON	NU	02	01	
22	TM2_M_DRWN	NU	02	01	
23	TM2_M_LOST	NU	02	01	
24	TM2_COACH	TX	15	01	
25	TM2_SCORE	NU	03	01	
26	TM2_FOULS	NU	02	01	
27	TM2_PLYR_NAME	TX	15	12	
28	TM2_PLYR_JNO	NU	02	12	
29	TM2_PLYR_POSN	TX	03	12	
30	TM2_PLYR_AGE	TX	02	12	
31	TM2_PLYR_HT	NU	02	12	
32	TM2_PLYR_WT	NU	03	12	
33	TM2_PLYR_SCR	NU	02	12	
34	TM2_PLYR_FLS	NU	01	12	
35	REF_NAMES	TX	12	03	
36	REF_COUNTR	TX	03	03	
36	VARIOUS_TX	TX	15	10	(Date, Venue,etc.)
37	VARIOUS_NU	NU	03	10	

BASKETBALL MAIN SELECTION AND ENTRY SCREEN



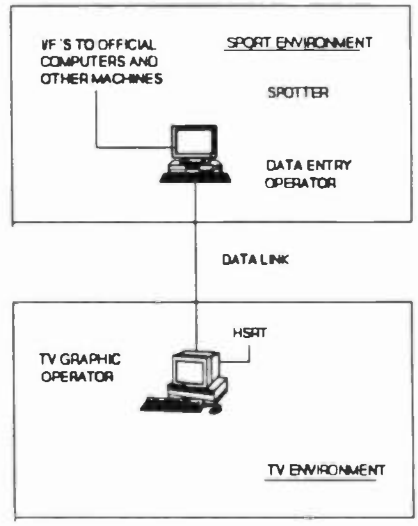
APPENDIX: I I I

SYSTEM SOLUTIONS



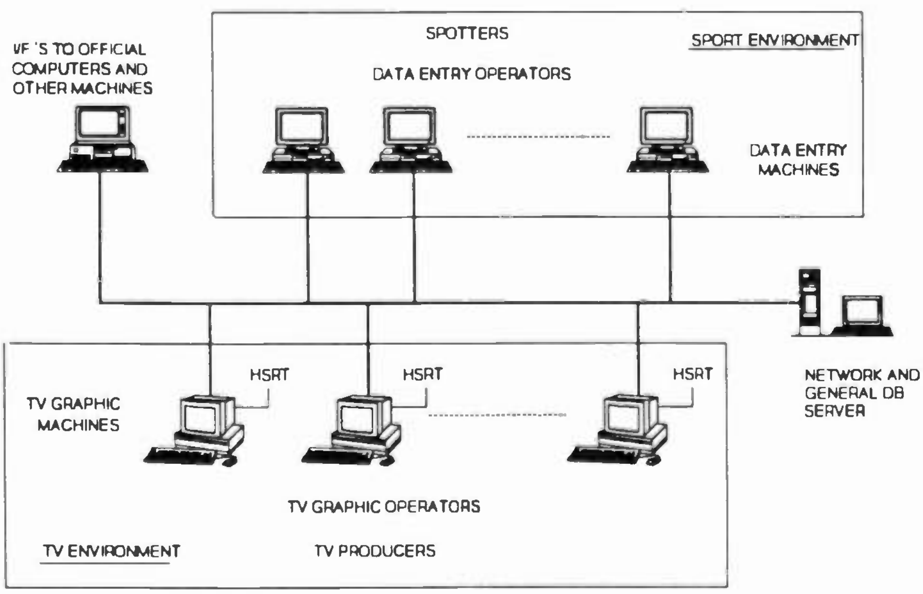
HSRT: HIGH SPEED RUNNING TIME

Fig 1: Stand alone system



HSRT: HIGH SPEED RUNNING TIME

Fig 2: System with external data entry



HSRT: HIGH SPEED RUNNING TIME

Fig 3: High End System

DIGITAL TELEVISION EFFECTS AND CONTROL

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The bridge between the different and sometimes competing worlds of video effects and computer-generated images is built with logic chips. It's fair to say that everyone on each side of this bridge understands the power of digital image processing and that one thing they all would agree on is that its technical rate of change is overwhelming at times.

Advanced LSI chips and microprocessors are essential to both worlds. In the computer, the control and manipulation of literally megabytes of data requires ever more sophisticated processors with greater speeds and capabilities for every new generation of the computer. In the video world, the demands for more sophisticated digital processing in the generation of video effects equates directly to speed and capability as well.

Computer graphics images are constructed pixel by pixel under microprocessor control, based on the human input of a graphic artist. RAM capacity, operational speed and the digital architecture, which dictates pixel quantization and organization, are all-important parameters in such computer systems.

In the world of video, the generation of images generally starts with analog signals which are converted to the digital domain. At this point, the video pixel representation is, however, no different from that of computer graphic signals, except perhaps for the detailed architecture. Further, state of the art computer graphics and video require essentially the same considerations with respect to RAM, speed of processing, and pixel quantization levels.

This paper will describe in block diagram form, the processes that generate digitized video signals. Included will be the overall processing architecture, the capabilities current systems can provide and how they are controlled to create video effects. Parallels will be drawn between the computer and video image processing. A peek into the possible future of video effects will be included.

The basis for the video effects systems described will be the FOR.A MULTIFEX product, which is now in its fourth generation of development and production.The bridge between the different and sometimes competing worlds of video effects and computer-generated images is built with logic chips. It's fair to say that everyone on each side of this bridge understands the power of digital image processing and that one thing they all would agree on is that its incredibly fast rate of technical change is overwhelming at times.

The purpose of this discussion is to explore the development and creation of 3D image effects in television-based systems, and compare them to such development in the Personal Computer. While there is a significant number of similarities closely linked both in architecture and in the extended use of advanced LSI chips, there are major differences as well. I would like to begin this comparison by looking at the background and evolution of Television digital effects and the manipulation techniques generally used in such equipment.

EVOLUTION OF DIGITAL TELEVISION EFFECTS

Digital television processing began decades ago with the digital TBC. Its basic architecture is shown in Fig. 1. I'm certain this figure is familiar to most of you. The sampling frequency most commonly used was 4sc and 8-bits, resulting in a 14.32 MBytes data rate through the memory. Initially, the memory capacity was relatively small -- only a few H-lines -- but after a few years it advanced to a full television frame. The digital storage required for this 14.32 MBPS data rate is 478 kBytes. The full fame TBC and counterpart frame synchronizer provided two advantages over smaller range TBC predecessors. The first advantage was that they could process non-synchronous VTR sources, and the second was freeze-frame aspect. Although this was the first and most simple of video effects, it was also important in handling non-synchronous, "hot" source switches, or when the input was not continuous.

The development of digital effects equipment followed two paths from this initial concept. One path led to the broadcast higher-performance end; the other to the lower-end B&I/prosumer level, based on the full-frame TBC product just discussed.

Once digitized, the pixel samples could be easily manipulated to create a number of effects. Such manipulation added no perceivable distortion to the picture, with the exception of compression and expansion, which could introduce alias frequencies. Examples of these simpler effects included *mosaic*, in which pixels were repeated horizontally and vertically to form tile patterns; *paint*, where the chroma bits were reduced in number to create contours; and *mirror* effects where memory READ addresses and sequences were reversed to flip the picture top-to-bottom or left-to-right.

Most effects systems used a signal component processing architecture because it is virtually impossible to perform sophisticated digital effects otherwise. This processing architecture is shown in Fig. 2. In this example, digital component video is generated using three A/D convertors; one each for Y, P_B, and P_R. As per CCIR Rec. 601, Y is encoded to 8-bits at 13.5 MHz, and P_B and P_R are each encoded to 8-bits at 6.75 MHz. The combined data rate is 27 MBytes.

Today's 3D digital effects processing is totally dependent on sophisticated microprocessor hardware and software and has a strong resemblance to the modern PC architecture, drawing many parallels with computer operation. Fig. 3 is a block diagram showing the 3D processing of the FOR.A MF-3000 Multifex unit. The major functional blocks in this diagram include the sub-effects generator, anti-alias filter, frame memory, interpolation circuit, 16-bit 68000 series CPU, serial interface, 3D manipulator, page-turn address generator, and key signal generator.

The majority of the above networks are comprised of gate array chips. Initial systems parameters are stored in the ROM and downloaded to RAM when the system is turned on. The Sub-effects section generates a variety of different 2D effects including mosaic, paint, and image negative. This is followed by the anti-alias frequency filter to band-limit the signal proportionately with the compression parameters in order to avoid introducing picture disturbances commonly referred to as "jaggies." The resulting signals are then written into the effects frame memory. It is here that the 3D effects are generated by controlling the H- and V-READ addresses. Note that the system includes both a 3D manipulator and a page-turn address generator. Both create the 3D effects in the system. Following the effects memory, a four-point interpolator circuit processes the signal to produce the final output.

Calculations for all processing are performed by the 16-bit CPU, based on command data from the operation control panel which enters the system through a serial I/O controlled by a Z80 co-processor. It is then converted to parallel and read by the CPU. The bus buffer connects commands to either the 3D manipulator or the page-turn address sections, which in turn generate H- and V-READ addresses to control data read out from the frame memory shown in the figure. The resulting H and V addresses are also used to generate key signals to be used to key the effects picture into another video path.

Fig. 4 shows the overall block diagram of the system including all interfaces. This diagram is for a full-option, dual channel effects system. The input processing units decode composite inputs and convert them to 4:2:2 component digital signals. There is, of course, no need to decode component analog inputs. There is a frame memory TBC function included in the processor. The digital outputs are connected to the effects board, which includes the effects memory, microprocessor control elements and the addressing circuitry as described in the previous figure. The output unit combines the digital outputs from the effects units and includes an image trail function which adds trail and "sparkle" effects using recursive delay circuits. The assembly also includes the D/A convertors to change the digital format back to analog. The video I/O board includes all of the input and output buffer amplifiers, and by-passing circuit functions. It also includes the interfaces for the gen-lock, DOC inputs, and background video input, and interfaces the control panel operation unit.

The effects provided by such a system are calculated from rather complex matrix equations. Fig. 5 is a representation of a 3D model, showing the viewing point and the viewing plane looking at a three-dimensional object. Without getting into the complexities of this model, we can look at the viewing plane as two-dimensional, with the 3D object and its motion projected onto it.

The plane represents the resulting output picture. The system functions to read the picture from the effects memory and control the H and V addresses to create the projected image on the plane as a result of a moving 3D object. For some functions, such as address control, it would be easier to modify the image when it is written into memory.

Other functions, such as interpolation and signal expansion, are easier to do on the READ address side. However, the MF-3000 does all of the effects on the READ side of the memory in this system. Some manufacturers do control the WRITE addresses to create effects. The most difficult effect to do on the WRITE side is expansion because the WRITE clock frequency must be increased in proportion to the expansion ratio. Fig. 6 is a table that relates the effects capabilities of the MF-3000, the type of effect and the basic implementation.

THE GROWTH OF THE DIGITAL PC

Digital television signal processing and the PC grew in parallel PC. While Apple started it all in the late 1970s, IBM launched a PC in the early 1980s that operated at a 4.77 MHz clock with an 8-bit bus architecture. The RAM space was only 64 Bytes. Today most of us are somewhat perplexed whether to go with 25 or 33 MHz or even the choice between 50 and 66 MHz. 32-bit architecture is the way to go, it's clear -- well, clear for today anyway -- and 16 MBytes of RAM is enough. Right?

Let's look at the evolution of the architecture from the beginning, up to the new and most advanced computers designed around the newer Intel and Motorola CPU devices.

Fig. 7 shows the system architecture used by the current generations of IBM-compatible PCs. Note that most of the peripheral control is on Industry Standard Architecture (ISA) or EISA bus. We also note that the interface functions are usually expansion boards connected to the bus.

These single CPU computers run at 15-30 MIPS, but will advance to as fast as 70 MIPS in the near future. They are fast, certainly in comparison to the first PCs, but when we consider them in light of the rates at which we need to process digital video, there's a real problem. Even with EISA 32-bit architecture operating at 33 MHz these computers cannot come close to handling digital video data rates in real time. Even a peak bandwidth of 132 MBytes is not nearly enough processing time to support the sustained data rate of 27 MBytes, which is necessary to process digital component video in real time.

For example, we have calculated the time to convert the addresses in the MF-3000 effects equipment to carry out a 3D move. At 30 MIPS, it would take 20 seconds. Not even close to real time processing! Of course, in all fairness to the PC, it was never intended to process live color video. It was a black and white text-functional architecture from the start and no degree of supercharging it will lead to the results we seek in the computer-video development. We must seek an alternative -- one that has in fact already been developed, and used in sophisticated 3D graphics work stations. These faster systems can handle the needed speed, have integrated high-performance graphics, built-in networking and mass storage control.

Fig. 8 relates an advanced PC and is exemplary of most work stations in use today. It has a two-tier bus structure. One is a wide, limited-length, high-speed bus for speed-critical operations and the other is slower and intended for I/O functions such as those shown in the diagram. It's interesting to note that the higher-end PC and the work station architecture are converging in basic structure, as we note in this figure. We should also acknowledge the contribution that "Windows" and substantially-increased RAM capacity have played in advancing the PC toward the big computer work station. These achievements are recognized not only on their technical merit but also for maintaining a low cost structure that offers sophisticated computer power to a wide range and large number of end-users.

Fig. 9 is similar to Fig. 8 and shows the block diagram for the Intel Peripheral Component Interconnect (PCI) local bus system. PCI is a standard that allows designers to interface directly with the higher speed CPU bus to be used for those peripherals requiring such. It also provides for an interface expansion bus chip, set to interface with a standard (and slower) I/O bus.

In a recent article Mr. Jim Clark, founder of Silicon Graphics computers, relates the history and development of graphics images. He notes that computer graphics technology is nearly 30 years old, having started at MIT in the early 1960s. Initially, using an analog computer, 2D images were painstakingly rendered by computer graphics pioneers such as Ivan Sutherland. As the field grew, greatly aided by the growing sophistication of digital computers, the techniques were developed to generate 3D images.

3D graphics generation can be considered as two distinct techniques: photo-realism graphics and real-time graphics. Photo-realistic graphics are not generated in real time and are generally intended for still or animation sequence applications and purposes. Graphics pioneers found ways of constructing 3D images from polygons and later from "patches" that formed smooth surfaces. These building blocks could be fabricated down to the size of a pixel. Later, lighting effects were added and produced incredibly striking images. It takes a relatively long time to render photo-realistic graphics, even with the modern super-computers we have today. In the future, however, it may be possible to render them in real time.

Real time 3D graphics are vastly different from photo-realism graphics. They are interactive and operate in real-time allowing, for example, designers of new products to model and look at assembly line aspects and identify problems in manufacture before the first prototype is even built. Flight simulators and other such instructional interactive applications are, of course, well established uses for real-time 3D graphics.

Clark notes that the cost of 3D graphics-generation hardware is relatively expensive. However, he points out that the marriage of the computer and video worlds makes economic sense if we look at what is now possible in the HDTV arena. Many of the processes that HDTV transmission requires, including compression, are being developed and are not far from completion as practical and economically viable systems. Another point is that the texture-mapping of images could be done with images captured using HDTV hardware, digitized and stored in the computer, rather than made from scratch. Such is currently possible using the system shown in the block diagram in Fig. 10. This is a practical system that exists today and another example of the emergence of the integrated combination of the world of the computer and video. The product is called the HMC-1020 Still Image Capture System.

COMPARISON OF 3D VIDEO PROCESSING AND COMPUTER-GENERATED IMAGES.

If we compare the digital video systems in use today to create 3D effects, with PC 3D graphics generation, it's clear there are many identical processes. Both require a CPU, RAM, ROM, and mass storage and display interfaces. They differ in two fundamental ways; the first is in where the input source comes from. In the case of video, it comes from a camera or perhaps a graphics/character generator. In the case of 3D graphics, it originates from within the computer, with the exception of the Image Capture System just described. Secondly, major differences exist today between the processing speed of the PC and that of the video processor. But both are changing, as we have just noted, and the bridge between video and computer-generated graphics will shrink, and in all likelihood disappear, in the not-too-distant future.

A LOOK INTO THE FUTURE

It's anyone's guess where the PC computer speed-race will go. As we have noted, however, it's not the speed of the CPU that makes the difference as far as processing digital in real time -- it's the architecture and how the CPU speed can be coupled to the bus that controls the video. Are we going to see 1,000 billion instructions in the future? Many think we will by 1995. Are we going to see the PC become the center of multimedia in our homes before the Year 2,000? Many also think so and share the expectation that even more realistic 3D graphics, rendered at a fraction of the time and cost required today, will be seen in our homes as a part of a highly sophisticated communications system that is totally based on a computer. Just looking at how far we have come in the past 10 years -- even in the last five -- I believe we can expect breakthroughs that we cannot even imagine today. It is an exciting time; better fasten your seat belt and let's hope it's a smooth journey -- simulated by 3D graphics and effects or otherwise.

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INTERNATIONAL BROADCASTING

Monday, April 19, 1993

Moderator:
Harvey Arnold, University of North Carolina, Center for Public Television, Research Triangle Park, North Carolina

A VERY HIGH POWER MULTI-CHANNEL VHF-UHF TV INSTALLATION IN THE ARABIAN GULF
Todd R. Loney and George R. Townsend
Micro Communications, Inc.
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A 7-GHZ BANO SUBMINIATURE FPU TRANSMITTER
Hideo Mitsumoto, Kazuo Imai, Masaru Fujita and Takao Murata
NHK Science and Technical Research Laboratories
Tokyo, Japan

EXAMINATION OF A PROGRESSIVE COMPONENT TV SYSTEM WITH AN EYE TO MEOIA CONVERSION
Masakatsu Tanaka, Tadao Kurosaki, Akihiro Hori, Masayuki Ishida, and Keiichi Saji
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Tokyo, Japan

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Phillips TV Test Equipment A/S
Broendby, Denmark

***PROGRAM DELIVERY STRATEGIES IN THE 1990's**
Kim Andrew Elliott
VOA
Washington, District of Columbia

*Paper not available at the time of publication.

A VERY HIGH POWER MULTI-CHANNEL VHF-UHF TV INSTALLATION IN THE ARABIAN GULF

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INTRODUCTION

With the end of the Arabian Gulf war in February of 1991, the Ministry of Information in Kuwait was faced with the difficult task of reestablishing radio and television broadcasting service. The facilities existing prior to the war had either been damaged, destroyed, or "requisitioned" for operation in Iraq. MCI and its agent, the S. H. Behbehani and Sons Company, were awarded the contracts to provide equipment, installation services and civil works to rebuild the broadcast facilities. The first effort was to construct an AM, an FM and two 6 kilowatt VHF stations on a rapid deployable basis. The war ended on February 27, 1991, and the stations were all on the air by June 10, 1991. These installations established acceptable service and thus provided a longer period of time to construct the major facilities.

When the war started, a major television transmitter facility was in the initial stages of construction on Failaka Island, located 18 miles offshore from Kuwait City in the Arabian Gulf. After the end of the war, the design of the Failaka Island facility was modified and upgraded in a joint undertaking by MCI and the Ministry of Information engineering department.

This paper describes the design and the unusual installation problems of the multi-channel 1600 kilowatt VHF and the multi-channel 5 megawatt UHF flagship transmitter stations.

PREPARATION OF THE FAILAKA SITE

The Failaka Island facility was originally planned in 1990 with Varian TVT in Cambridge, England doing the custom design work for the system. Construction of the large transmitter building was about three-quarters completed when the war started. The island was strategically located and hence the Iraqis heavily mined and constructed barricades and fortifications in anticipation of an invasion by U.N. forces. As a result, the incomplete transmitter building was damaged by a direct hit from a 500-pound bomb and from other lesser ordnance. No equipment had arrived at the site, but a 350 meter tower had been damaged to the extent that it had to be "dropped."

The extent of mine fields around the site was unknown, so the immediate area had to be probed and cleared before reconstruction work could commence. Failaka Island, as a result of the extensive damage and mining, was a restricted area and only personnel associated with the project were allowed access.

Dropping the guyed tower posed a real problem inasmuch as its base was only 35 meters from the front of the building. However, this was successfully accomplished with demolition charges placed at the guy anchors, and the tower fell, as predicted, away from the building.

During the reconstruction of the building, procurement of the equipment was taking place, some of which was stored in warehouses in Kuwait City with the larger items stored in cargo containers outside of the building at the site. Upon near completion of the building, MCI field engineers arrived on site in July of 1992 to mobilize the equipment within the building and to start installation of the entire system. At that time of year, ambient temperatures reach 120 degrees Fahrenheit with high humidity. The building air conditioning, rated at 400 tons, was not installed until several months later, thereby presenting a very difficult working situation.

High winds up to 50 miles per hour with consequent sandstorms represented an additional handicap to the outside work. The tower construction was delayed due to unusual rains, which turned the site into mud two feet deep.

The installation engineers were assisted by a number of local technicians and laborers working for the Behbehani Company. The language barrier was not too restrictive inasmuch as some of the construction supervisors spoke English.

SYSTEM DESIGN

The goal of the project was to construct two high power VHF and two high power UHF transmission systems that would establish service contours equivalent to Grade B to almost all of the populated area of Kuwait and to some of the abutting areas of Iran and Iraq. Engineering studies had determined that the VHF stations operating on channels 8 and 10 in Band III (high channel VHF) required an effective radiated power of 1.6 megawatts each, using a transmitter power of 40 kilowatts and that the UHF stations on channels 24 and 39 in Band IV required an effective radiated power of 5 megawatts each, using a transmitter power of 120 kilowatts. This represents a total VHF effective radiated power of 3.2 megawatts and a total UHF effective radiated power of 10 megawatts.

The site on Failaka Island is only 40 feet in elevation above sea level, so the system design required radiation centers for the antenna in the order of 1000 feet. A single tower was preferred, which dictated that the antenna system for the four channels must be limited to a reasonable overall length so that the centers of radiation would be high on the tower. The best way to address this criteria was to employ a single antenna for the two VHF stations and likewise for the two UHF stations. This also would guarantee that the radiated patterns would be identical for the two VHF stations and identical for the two UHF stations. Both the VHF and UHF antenna had to cover the full band, since it was planned to add additional channels later. Another requirement was for redundancy in the two antenna systems, and this was fulfilled by employing split feed antennas.

The passive systems from the transmitters to the split feed antennas required dual transmission lines, channel combiners, power splitters, switchless combiners and transmitter combiners and individual diplexers, since the transmitters for all channels employ dual operating in parallel. This overall passive system is doubtless the most complex system ever installed in a single facility.

The UHF system is all waveguide and is hung from the ceiling of the transmitter room area and occupies much of the overhead area. The VHF passive system is considerably smaller and is floor mounted, as shown in the transmitter room floor layout in Figure I. All four transmitter systems can be directly viewed from the central control desk, which contains all appropriate monitoring and control functions.

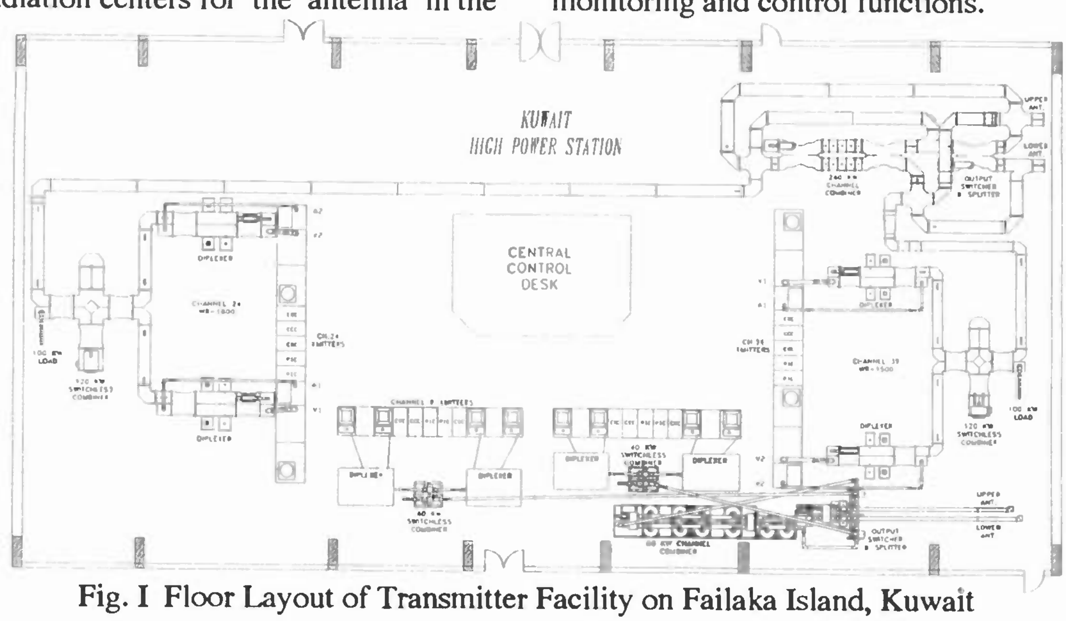

Fig. I Floor Layout of Transmitter Facility on Failaka Island, Kuwait

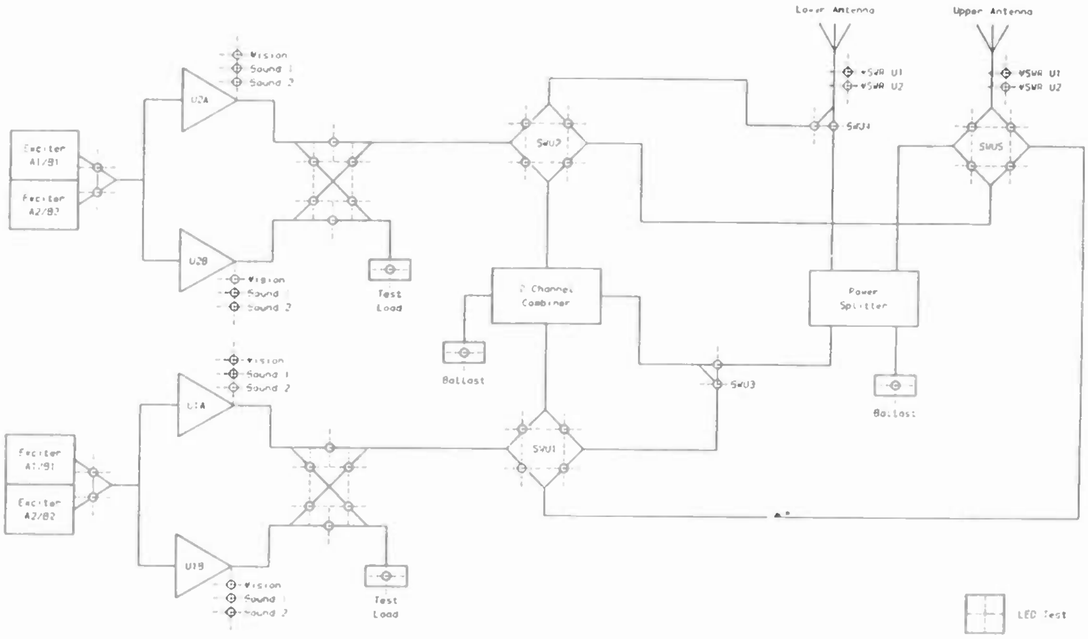

Fig. II RF Mimic Panel Layout

The central control desk also contains RF mimic panels that show precisely the status of each transmitter and all transfer switches. Figure II is an RF mimic panel layout. Control of the transmitters, the RF switch system and the mode selection are all available at the central control desk. Telemetering of all important transmitter readings and other elements of the system are provided.

Referring again to Figure I, it is seen that standard equipment racks are located between each of the transmitters making up a channel. The equipment in these racks are program input equipment, test and monitoring equipment, exciters and their changeover switching. All of this equipment is duplicated for each transmitter so that each transmitter system is independent of the others.

TRANSMITTER EQUIPMENT

The Ministry of Information specifications called for an exceptionally redundant transmitter system to perform under almost any emergency condition. Each channel was therefore comprised of two transmitters operating in parallel into a switchless combiner with automatic switching in the event of failure of either transmitter. Each transmitter, in turn, employs dual exciters set up for automatic/manual changeover.

An additional dimension of reliability was obtained for the VHF channels by the selection of Larcan all solid state 20 kilowatt, air cooled transmitters, which when paralleled for each channel, provide 40 kilowatts of peak vision output power.

The air cooling system for the transmitters was custom engineered to combat the high level of dust inherent with the sandstorms that are part of the weather pattern in the area. Power measurement of the VHF transmitters is with a water load and associated water heat exchanger.

The achievement of high reliability for the UHF transmitters was accomplished by the selection of 60 kilowatt klystron transmitters manufactured by Varian TVT in Cambridge, England. This model of transmitter has considerable history of high performance and high reliability operation. Two of the transmitters are paralleled for each UHF channel and thus provide 120 kilowatts of peak vision output power. The transmitters employ vapor cooling with two heat exchangers installed in an isolated area, which is elaborately constructed with an air filtering system to minimize the dust problem.

All four systems were specially designed for operation in dual sound mode (stereo/mono/dual sound) in conformance with the IRT German standard. When in dual sound mode, the two sound channels are usually used for two languages; e.g., one may be in the Arabic language and the other in the English language. Selection of the mode is a switching operation at the central control desk.

PASSIVE SYSTEM DESIGN

The passive systems requirements posed several challenges to the MCI engineering group.

Temperature Differentials in Transmission Lines

The split antennas required dual transmission line, so it was necessary to conduct a study as to the potential problem of phase changes between the lines due to any temperature differentials that might occur. Such phase changes at the input to the antennas could result in beam tilt changes. Studies had previously been conducted on a dual wave- guide installation made by MCI in Mutlaa, Kuwait, prior to the Gulf war. Although there was some differential in line lengths due to uneven warming of the two waveguide runs, it was not enough to change the phase relationship arriving at the split antenna to the extent that it significantly changed the beam tilt. Due to the much lower frequencies of the VHF line length, variations due to temperature variations are not a factor.

Waveguide Broadbanding

The waveguides feeding the UHF split antennas had to be capable of faithfully passing channels 24 and 39, representing a bandwidth requirement of 144 MHz. This translates to a 23% bandwidth, which is in sharp contrast to the normal single channel or 1% bandwidth required in stations in the United States.

Some of the broadbanding solutions were to minimize the number of flanges which cause additive VSWR problems over the 150 MHz bandwidth and by developing a special flange tuner to tune each flange. It should be noted that some of the custom engineering developments for broadbanding components for this project will be useful in the future for HDTV applications.

Combining Two Channels Into One Antenna

The combining of channels 8 and 10 for a total peak power output of 80 kilowatts required precision design parameters. Constant impedance band notch design was employed for this purpose.

The total of 240 kilowatts of UHF waveguide combining was well within the expertise of MCI engineering. Constant impedance bandpass design was employed in this instance.

The requirement to combine two 120 kilowatt UHF channels 144 MHz apart presented a new requirement for waveguide components. Components including transitions, hybrids, combiners, switches, etc., were all completely redesigned to be broadband over this much larger band than the typical 8 MHz channel width while limiting insertion loss to less than 0.1dB. MCI spent five to six months in extensive R&D, which resulted in components which could ultimately be used over a bandwidth of 150 MHz.

PASSIVE SYSTEM CONTROL

Remote Control of High Power RF Switches

Since the specifications called for remote control of all high power switches and switchless combiners from the central control desk, a microprocessor system based upon previous MCI designs, was developed. The microprocessor controls all RF system functions into one unit, which is operated from the central control desk. It monitors the status of the RF systems components, reads the current operating mode selected and determines whether the components are in their correct positions. For some switching operations, the microprocessor momentarily deactivates the transmitter until the switch is completed whereupon the transmitter is auto- matically turned back on. The only task for the operator is to select the desired mode of operation.

Solution of Problem of Multiple Frequencies in the Line after the Combiners

Monitoring and control after the combiners posed problems due to there being multiple frequencies appearing at the directional couplers. Previous designs for monitoring and control under circumstances of multi-channel frequencies present at the directional couplers caused inaccurate readings of VSWR. To overcome this, TVT designed a superheterodyne type receive device for coupling at appropriate points in the passive system. A derivative voltage from the receive device is fed to the VSWR metering. This allows each frequency to be monitored separately and totally independently of the other frequencies appearing at each directional coupler.

TOWER AND ANTENNA SYSTEM

The antennas had to be designed for all channels in the VHF band and all channels in the UHF band. The overall height of the Stainless guyed tower is 1044 feet. Design parameters for the antenna system specified:

- A top mounted UHF panel antenna
- A VHF panel antenna immediately below the UHF antenna
- Two WR1800 waveguide lines
- Two 5-inch heliax lines
- Several parabolic dishes
- An elevator
- Antennas and associated transmission lines for future FM, VHF and UHF facilities, one VHF, one FM and two UHF future antennas are planned and will be mounted at lower heights on the tower.

The tower also had to be designed to survive winds of up to 120 miles per hour and ambient temperatures of 70 degrees centigrade. Figure III is a plot of the tower and the antennas mounted thereon.

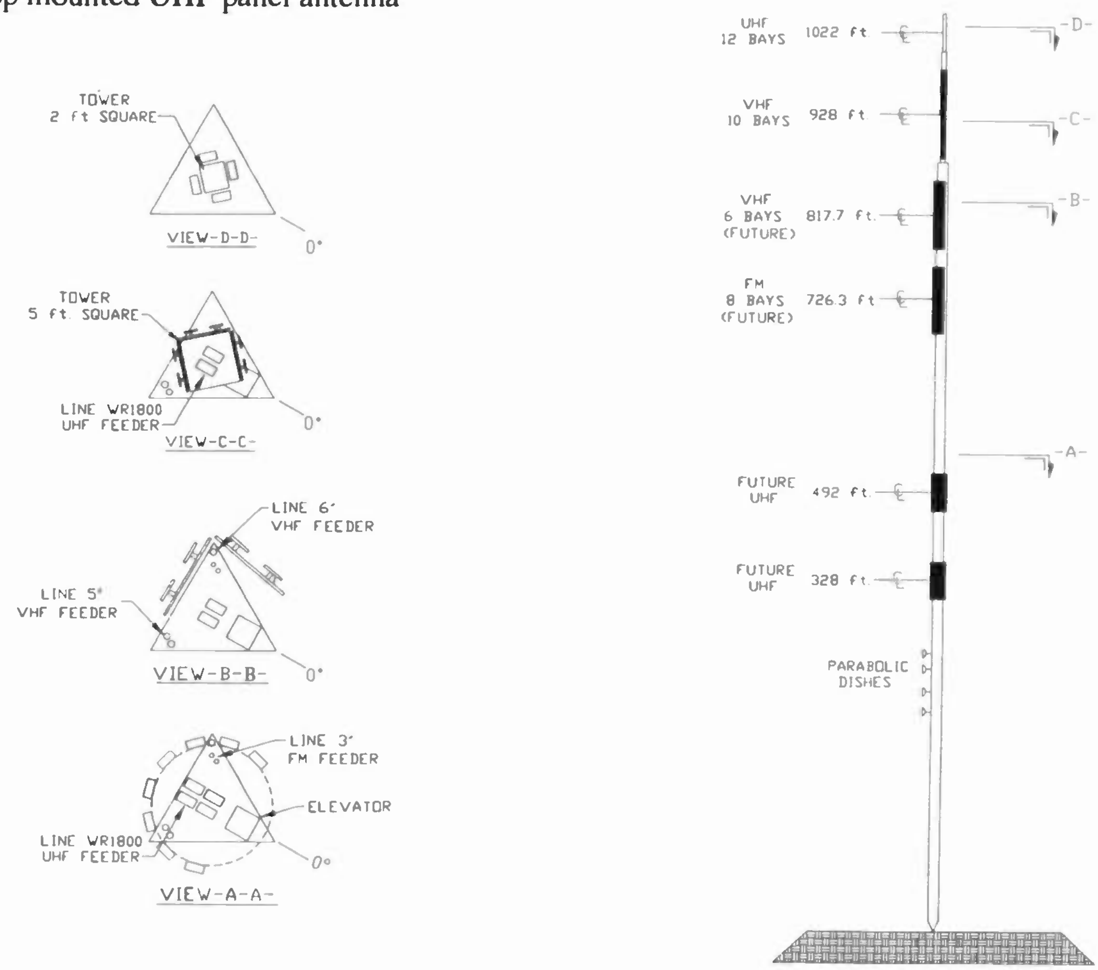

Fig. III Tower, Antenna and Feeder System

The detailed tower specification is shown in Table I.

SPECIFICATION

VHF SPLIT FEED ANTENNA BAND III	
PATTERN	WIDE CARDIOID
NO. OF BAYS	10
PANELS/BAY	3
GAIN-PEAK	40
TRANS. POWER	40kW/CH
EFFECTIVE WIND AREA	200 FT.^{2}
WEIGHT	5160 LBS.
UHF SPLIT FEED ANTENNA BAND VI/V	
PATTERN	OMNI
NO. OF BAYS	12
PANELS/BAY	4
GAIN-PEAK	40
TRANS. POWER	120kW/CH
EFFECTIVE WIND AREA	60 FT.^{2}
WEIGHT	8000 LBS.
VHF SINGLE FEED ANTENNA (FUTURE)	
CHANNEL	8/10
NO. OF BAYS	6
PANELS/BAY	2
GAIN-PEAK	15
FM SPLIT FEED ANTENNA (FUTURE)	
NO. OF BAYS	8
PANELS/BAY	2
GAIN-PEAK	4
UHF ALTERNATE ANTENNA BAND IV/V (FUTURE)	
PATTERN	CARDIOID
NO. OF BAYS	12
PANELS/BAY	8
GAIN-PEAK	54/58
CHANNELS	CCIR 24 & 39
ERP	4500 kW/5020kW
HAAT	460 FT
TRANS. POWER	120 kW/CH
EFFECTIVE WIND AREA	364.5 SQ. FT.
MICROWAVE DISH	
(2) 3 M	
(2) 1.2 M	

Table I Detailed Tower Specification

Panel antennas were chosen and custom designed for both the VHF and UHF systems for several reasons:

- Panel antennas are electrically and mechanically most suitable for split operation.
- Directional and tilt characteristics can be readily modified any time after installation.
- They can be easily serviced in place.
- They are readily stacked - as required in this application.
- They are broadbanded over the entire VHF and UHF bands as also required for this project.

Both antennas provided by MCI and designed by Sira, are split for dual feed operation. The top mounted UHF antenna consists of a 12 bay array with 4 panels per bay and a gain of 49. Each panel consists of 4 full wave dipoles fitted to a stainless steel reflector. The active elements and connectors are tin or silver plated copper and brass. Non-metallic elements are made of teflon and polyethylene. These materials assure a high quality product and hence, high reliability.

Each panel is protected with sealed epoxy glass radomes which are pressurized up to the dipole feed points. In this installation, the epoxy glass radomes serve as a good protection against the sand blast effect of the desert sandstorms. The panels are mounted upon four sides of a square section of tower top measuring two feet on a side. This dimension is critical since it provides an omni directional circularity of ± 2 dB for any channel in the UHF band, while allowing sufficient space within the structure for power splitters and flexible coaxial feeder cables serving the individual panels in addition to space for access by service personnel.

The VHF antenna has an aperture of 10 bays with 3 panels per bay. This results in a wide cardioid pattern with a peak gain of 40.

Each panel of the VHF antenna has a grid reflector to reduce wind load and to which is mounted four full wave horizontal dipoles.

SERVICE AREA

The area being served by the facility is atypical since Failaka Island is sparsely populated and the nearest part of the mainland of Kuwait is 18 miles distant. Thus, Kuwait City and other population centers to be served are all beyond the 18 miles. Taking this into account, the transmitting antennas were designed for no null fill and no significant beam tilt. The terrain in the populated part of Kuwait is relatively flat so that theoretical contours are realistically achieved. Figure 4 and 5 are maps of the service contours produced by the Failaka stations.

During certain times of the year, these contours are greatly extended into other countries bordering the Gulf. This is due to the "ducting" phenomenon realized as a result of the extremely high humidity and air temperature over the waters of the Arabian Gulf. The "ducting" is the bending of high frequency radio waves so that they essentially follow the curvature of the earth over long distances. The path length depends upon the length of the water path, and in the case of the Arabian Gulf, several hundred miles of over-the-horizon transmission are realized. The future antennas may be employed to take advantage of the 'ducting' since the ideal duct may be at different heights at different times.

The authors witnessed this phenomenon of ducting first hand during a visit to a television sales room in Kuwait City in July of 1992. With standard receive antennas and receivers, stations located in Bahrain at 260 miles, Qatar at 300 miles, and Abu Dhabi at 530 miles were being received with good Grade B quality although there were occasional momentary fades.

Although this ducting effect greatly extends coverage beyond the normal theoretical, the downside is that it has the potential of causing interference to local stations from co-channel stations located at a great distance. This constitutes one justification for constructing an exceptionally high power facility inasmuch as it provides a very strong desired signal in the service area relative to the undesired signal.

ACKNOWLEDGMENTS

In addition to the oppressive heat and humidity, the lack of air conditioning, the sandstorms and the delays in building completion, MCI and other personnel had to contend with isolation on the island at times, due to high seas in the Arabian Gulf, which denied them the use of the small boat used for transportation between the island and the mainland.

Near the end of the construction period, the renewal of some hostilities between U.N. allies and the Iraqis created a certain amount of tension among some of the workers with cruise missles and Eagle, Tornado and Mirage Jets flying over the island. We appreciated the accuracy of the allied technology but worried about the accuracy of the scud missiles.

In spite of all these difficulties, it is a tribute to all those involved in the construction of the facility that the installation was made pretty much on schedule. The authors wish to also recognize the fine support given to the MCI project engineers by Larcan, TVT, Stainless, Behbehani and by other MCI personnel.

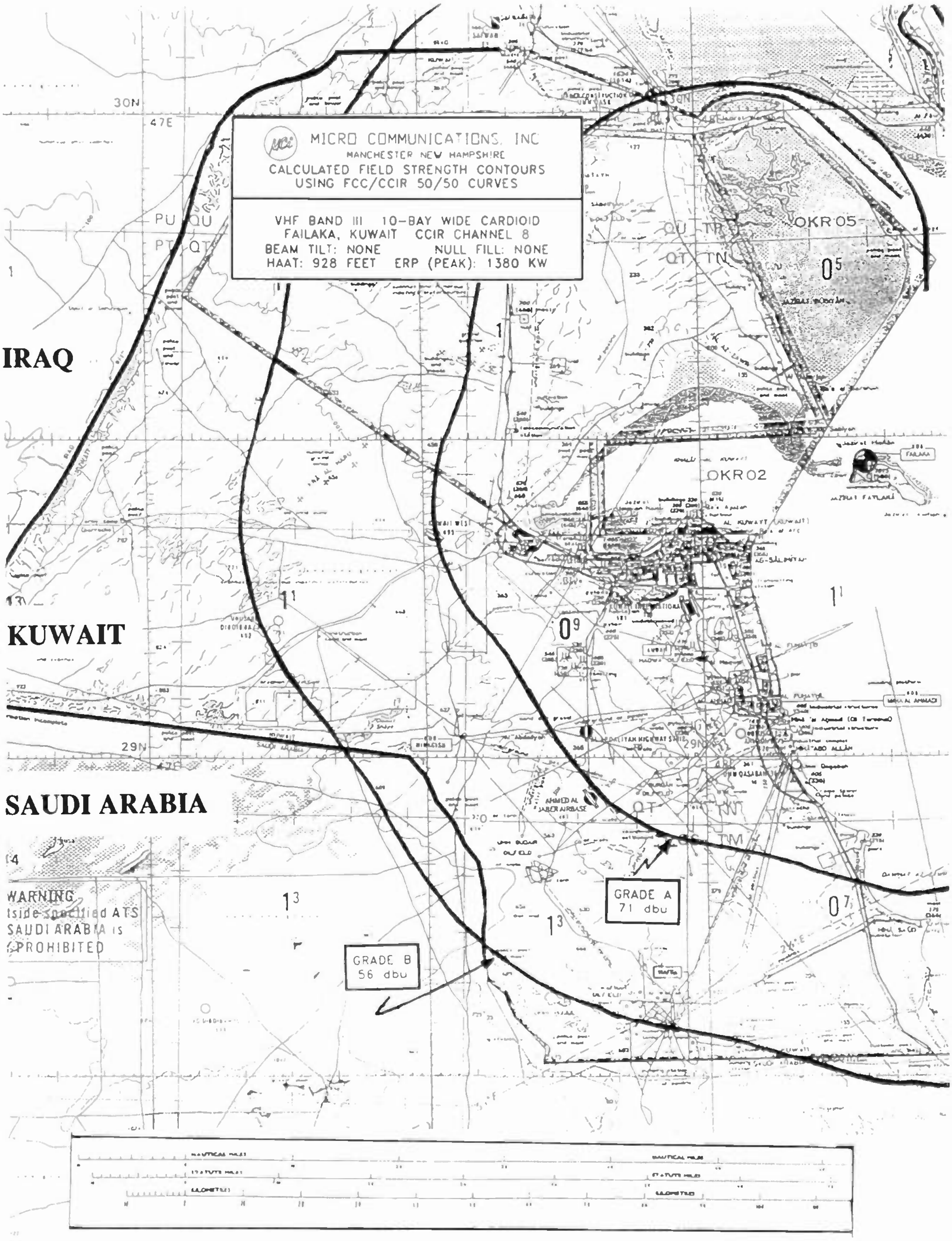

Fig. IV Calculated Field Strength Contours for VHF Band III

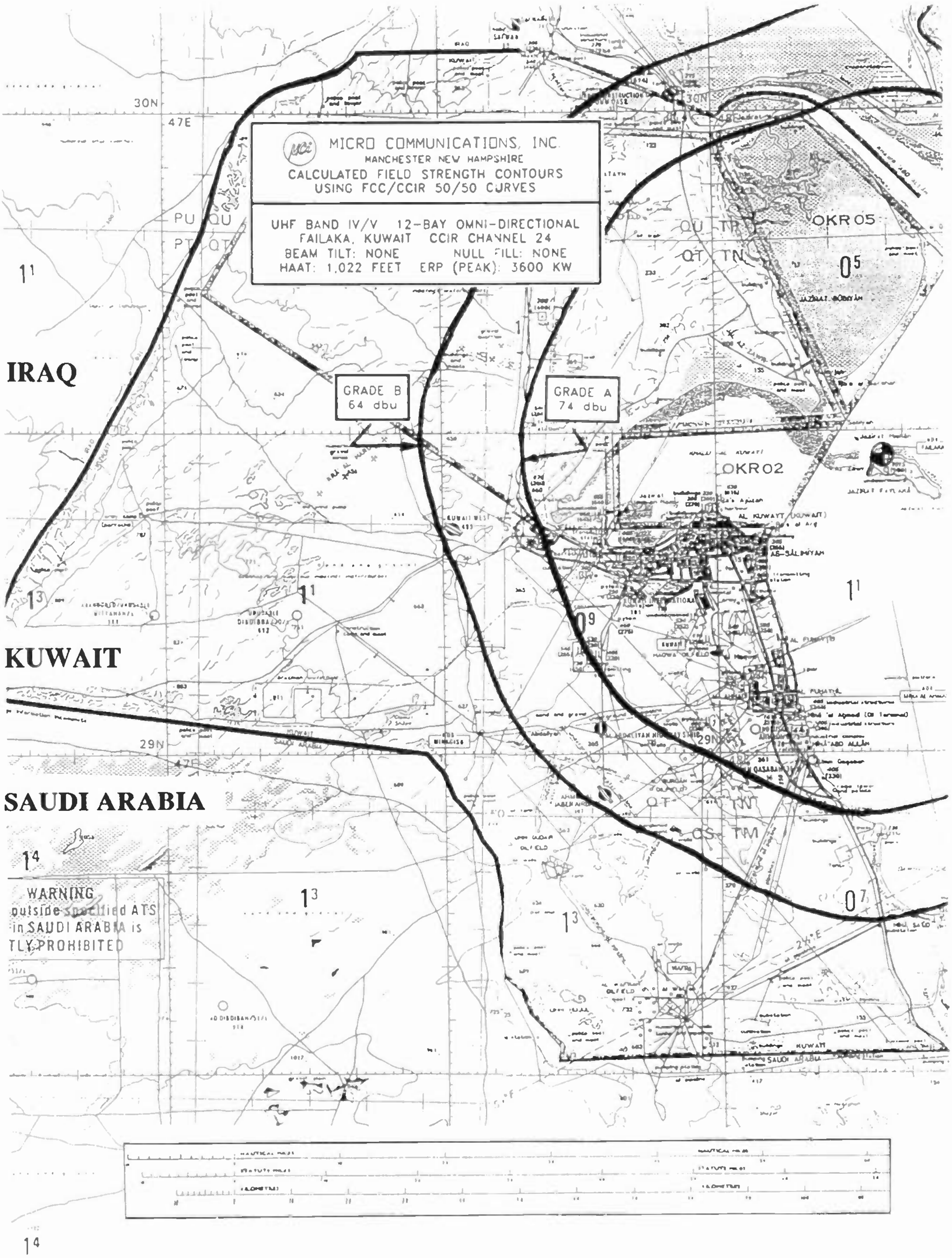

Fig. V Calculated Field Strength Contours for UHF Band IV/V

A 7-GHZ BAND SUBMINIATURE FPU TRANSMITTER

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ABSTRACT

The light, compact, low-cost FPU transmitter developed for NTSC TV signals and described here consists of a flat antenna and a 7-GHz-band FET direct frequency modulator. The modulator uses a medium-power microwave FET, a dielectric resonator, and two varactors. The microwave power of 160 mW is gained from only the FET. The use of two varactors magnetically coupled with the dielectric resonator results in good modulation linearity, so that there is no need for a nonlinear compensating circuit before the modulator. A differential gain of less than 3% and a differential phase of less than 2° are achieved. The 90×90-mm antenna consists of four rectangular microstrip patches printed on a Teflon-glass substrate, and it has an antenna gain of 13 dBi. The 60×45×90-mm transmitter weighs 300 g and has a power consumption of 2.2 W. When this transmitter is used with a receiver that has a noise figure of 4 dB and a parabolic antenna 60 cm in diameter, the transmission distance is about 10 km.

INTRODUCTION

After 200 years of dormancy, the Unzen volcano in Japan's Nagasaki prefecture began to erupt on November 17th of 1990. Since then, TV has reported the appearances of an eruption or other disaster site nearly every day. As a medium for transmitting program material from the field to a broadcast station, a field pick-up (FPU) microwave link contributes to the dynamic and on-the-spot coverage of these kinds of events. Its role in covering disasters is particularly important because an FPU can swiftly provide detailed information from remote sites.

The Unzen eruption stimulated NHK to begin developing equipment for gathering news more safely, equipment such as a remote-controlled cameras and solar batteries. The transmission equipment used for live coverage from dangerous locations must be small and light. Low power-consumption is also an important feature under conditions where power supplies may be inconvenient or uncertain or both. And equipment that may be lost in dangerous situations should be as inexpensive as possible.

A new FPU transmitter has therefore been designed to provide the minimum functions required for gathering emergency news and to be extremely compact, light and low cost. This paper describes the concepts, components, and performance of this transmitter.

DESIGN CONCEPTS

Because the conventional FPU unit was designed for general use, its capabilities include multichannel transmission and wide coverage. The size and weight of a typical conventional FPU transmitter are therefore 170 × 140 × 330 mm and 8 kg. The most important features of an FPU transmitter used for emergency news gathering, however, are compact size, light weight, and low cost. We therefore designed a transmitter that would provide only the following minimum functions:

1) 10-km transmission
2) one transmission channel in a 7-GHz band (6.4-7.1 GHz).
3) one video channel and one audio channel

Table 1 lists a sample link budget at 7 GHz. This budget includes a 3-dB margin for the attenuation caused by a heavy rainfall of 40 mm per hour. The table indicates that picture quality with a subjective grading of more than 4 (an unweighted S/N of 41.9 dB) can be obtained after 10-km transmission by using a transmission power of 22 dBm (160 mW) and a transmitting antenna with a gain of 13 dBi. The 160 mW can be obtained with a single medium-power FET, thereby reducing power consumption and cost by eliminating the need for a high-power FET. Low power consumption also lengthens the time for battery-powered operation. And by applying the second and third concepts, a sufficiently low cost and small size can be achieved.

TABLE 1: Example of link budget for 7 GHz band FPU.

Channel bandwidth	(MHz)	17	
Frequency deviation	(MHz)	8	
Base bandwidth	(MHz)	4.5	
Transmission power	(dBm)	22	160mW
Transmitting antenna gain	(dB)	13	
Free-space loss	(dB)	−129.3	10km distance
Miscellaneous loss *1	(dB)	−4	
Receiving antenna gain	(dB)	30	D=0 6mϕ
Noise temperature *2	(dBm)	−97.7	NF = 4dB
Total C/N	(dB)	29.4	
FM gain	(dB)	12.5	
Overall unweighted S/N	(dB)	41.9	

*1 Miscellaneous loss includes the rain attenuaion of 3dB for 40mm/H heavy rainfall.

*2 Noise temperature includes terrestrial noise temperature.

CONSTRUCTION OF FPU TRANSMITTER

To generate frequency-modulated microwave signals, techniques with a multiplier or a mixer are generally used: a frequency modulated intermediate frequency is mixed or multiplied up to a microwave frequency. Both those techniques, however, require many components and are not cost-effective. Another technique, and one requiring fewer components, uses a direct frequency modulator that directly generates a microwave modulated by a baseband signal. This kind of frequency modulator is suitable for a compact FPU transmitter.

A block diagram of the FPU transmitter is shown in Fig. 1. The video input signal is 1 Vp-p (input impedance = 75 Ω), the audio input signal is 0 dBm (input impedance = 600 Ω) and the dc power supply is 12 V.

After the video input signal passes through a low-pass filter (LPF) and a pre-emphasis circuit (EMP), it is amplified by a video amplifier. The audio input signal is frequency-modulated at 6.8 MHz by an audio modulator circuit whose frequency is stabilized by a phase lock loop (PLL). The modulated audio signal is multiplexed with the video signal and is input to a 7-GHz-band direct modulator. The frequency-modulated 7-GHz signal is led to a flat antenna via an isolator. Every part of the electronic circuits is miniaturized by using chip components and operates under single polar voltage of +10 V. The key devices of this transmitter are the flat antenna and the 7-GHz-band direct frequency modulator.

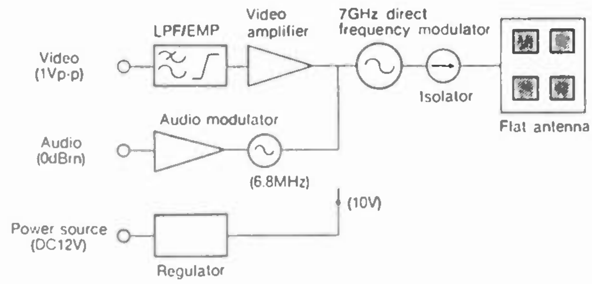

FIGURE 1: Block diagram of FPU transmitter.

7-GHz-BAND DIRECT FREQUENCY MODULATOR

This modulator (Fig. 2) consists of a common-drain GaAs FET oscillator stabilized by a dielectric resonator (DR) and two varactor diodes. The DR is magnetically coupled with a microstrip line connected to two hyper-abrupt junction varactor diodes, one on each end[1,2]. By varying the varactor capacitances with the applied video signal, the resonant frequency of the DR is changed and frequency modulation is accomplished. The configuration shown in Fig. 2 provides a higher modulation sensitivity than does a configuration using a single varactor[1].

The output power is extracted by a probe magnetically coupled with the DR. The DR acts as a band-pass filter, suppressing the second and the third harmonics generated in the FET to 100 μW or less. The circuit is fabricated on a 1-mm-thick alumina substrate with a relative dielectric constant of 10. A DR with an unloaded Q of 1000 is used.

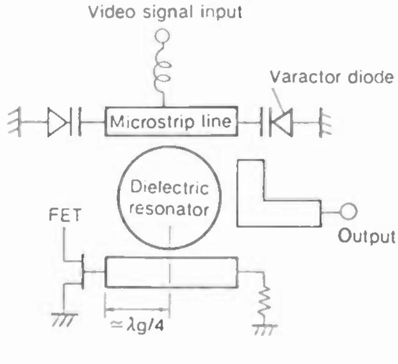

FIGURE 2: Construction of 7 GHz band frequencymodulator.

High modulation sensitivity generally allows use of low–gain video amplifier, and good modulation linearity allows to omit the nonlinear compensating circuit before a modulator. The output power of the modulator that is emitted directly should be as high as possible. And for accordance with radio regulations, frequency stability is very important. To provide these inconsistent characteristics, this modulator optimizes the combination of the modulation linearity, the output power, and the frequency stability .

The position of the DR and the coupling coefficients between the DR and microstrip lines were determined by using the equivalent circuit of varactors and DR and the measured negative resistance of the common–drain FET[3]. As shown in Fig. 2, the DR is located at about one–fourth of the guided wavelength away from the gate of the FET. The modulation sensitivity is set to 6 MHz/V to keep the oscillation frequency stable, and a DR (ε_r=38) with a resonant frequency coefficient +20 ppm/°C is used to compensate for the oscillation frequency drift.

The temperature dependencies of the modulator output power and oscillation frequency are shown in Fig. 3. The 74–ppm frequency drift over a temperature range of 0 to 40 °C is accepted by the radio regulation in Japan, and an output power of 21.8 dBm is obtained.

The temperature dependencies of the differential gain (DG) and differential phase (DP) are shown in Fig. 4. These measurements were made with a maximum frequency deviation of 8 MHz through the emphasis circuits and at average picture level (APL) ranging from 10% to 90%. The DG and DP shown in Fig. 4 are the minimum and maximum values measured when varying APL from 10% to 90%. A DG of less than 3% p–p and a DP of less than 0.2 deg. p–p show that this modulator achieves good DG and DP characteristics without using expensive nonlinear compensating circuits.

FLAT ANTENNA

A flat antenna is suitable for use with a compact FPU transmitter because it is light, thin and highly portable. A small and thin antenna avoids strong wind pressure and requires only simple mounting fixtures. And because it can be easily manufactured by etching, its cost can be low.

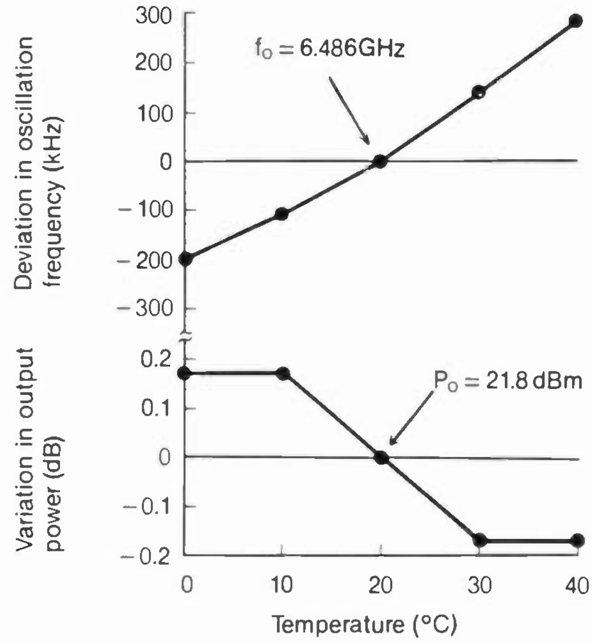

FIGURE 3: Temperature dependencies of oscillation frequency and output power.

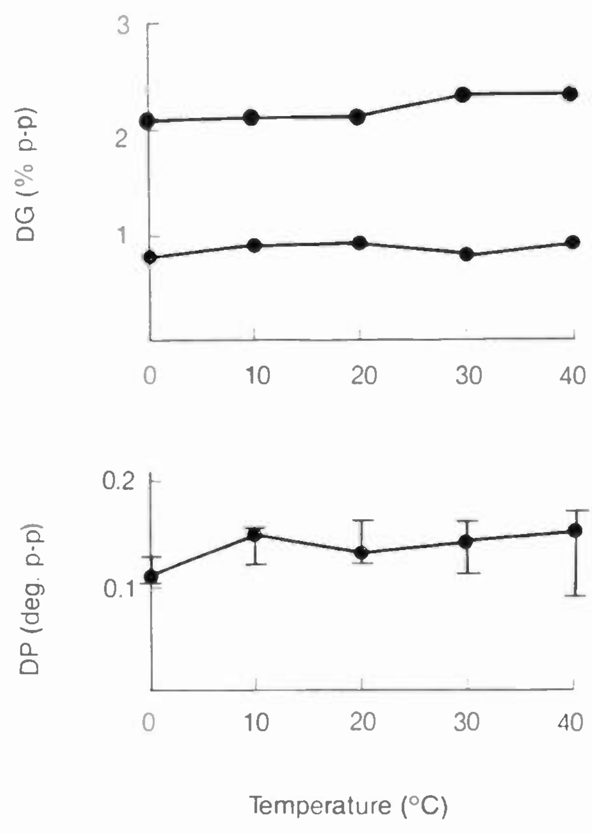

FIGURE 4:Temperature dependencies of differential gain (DG) and differential phase (DP).

The antenna (Fig. 5) is constructed of four rectangular microstrip patches to excite a linearly polarized wave, and its outer dimensions are 90 mm × 90 mm. For high gain, the spacing between the rectangular patches is 0.9 λ_0. The 1.6-mm-thick Teflon-glass substrate with a relative dielectric constant of 2.6 is highly resistant to corrosion, water, and heat.

The measured and calculated radiation patterns of the experimental antenna in the E-plane at 6.5 GHz are shown in Fig. 6. The measured pattern shows a good agreement with calculated data, and a half-power beamwidth is 29°. The bandwidth with a VSWR of less than 2.0 is 200 MHz and the gain is 13 dBi.

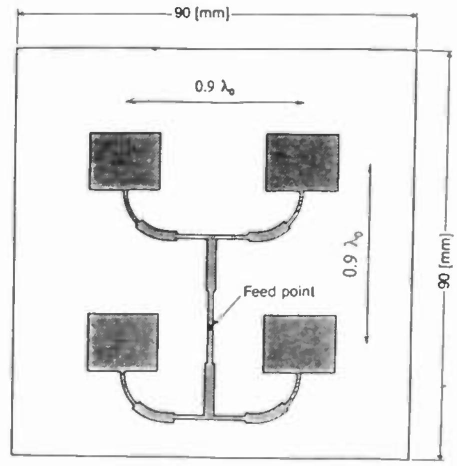

FIGURE 5: Schematic layout of 7 GHz band flat antenna.

PERFORMANCE OF THE 7-GHz-BAND FPU TRANSMITTER

An external view of the experimental FPU transmitter is shown in Fig. 7, and the transmitter performance at ambient temperatures between 0 and 40°C is summarized in Table 2. A transmitting power of 160 mW and a frequency stability within ±100 ppm is obtained. The DG is less than 3% and the DP is less than 2°. With a 12 V dc power supply, the power consumption is 2.2 W. This 60×45×90-mm transmitter weighs 300 g.

Figure 8 shows the external appearance of a practical transmitter. The substrate of the flat antenna is covered by a radome made of low-loss dielectric material. The antenna and transmitter are combined solidly in a body for easy operation. To transmit images, you have only to switch on the FPU transmitter. The electrical characteristics are the same as those of a trial FPU transmitter. Practical FPU transmitters are now in use for daily news and they work satisfactorily.

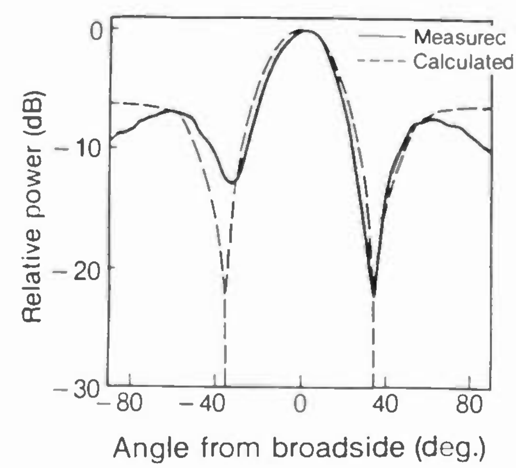

FIGURE 6: Radiation pattern of flat antenna in the E-plane at 6.5 GHz.

OPERATION OF A SUBMINIATURE FPU

Because of the compactness and low power consumption of this new FPU transmitter, it can be easily loaded on a remote-controlled helicopter along with a compact camera. This system can then capture images in dangerous areas and transmit them, via a relay base, to a broadcast station. And by using a video camera and a solar battery, this transmitter can be used for long-term observation of a dangerous spot, such as a volcano. This transmitter thus enables the assembly of compact remote-controlled camera systems. This transmitter is also available for short-range pickups such as golf matches, political conventions,

FIGURE 7: External view of FPU transmitter.

and parades.

CONCLUSION

A small, light-weight and low-cost FPU transmitter was developed. The key devices in this transmitter are a flat antenna and a 7-GHz-band direct frequency modulator. An output power of 160 mW is obtained from a single medium-power GaAs FET, and the oscillation frequency of the modulator is stabilized by a dielectric resonator without a phase lock electronics. Good modulation linearity is attained by a pair of hyper-abrupt junction varactors, eliminating the need for a nonlinear compensating circuit. The flat antenna consists of four microstrip patches printed on a thin and inexpensive dielectric substrate. The antenna's dimensions are 90 mm × 90 mm, and its gain is 13 dBi. The performance of this FPU transmitter rivals that of conventional FPU units, so new operational possibilities should be opened up by this compact high-performance FPU.

ACKNOWLEDGEMENTS

We thank Dr. Utsumi and Mr. Yazawa of NHK for their guidance and encouragement throughout the course of this work.

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TABLE 2: Performance of FPU.

Frequency	6.486 GHz
〔Transmitter〕	
Power output	160mW (22dBm)
Frequency stability	±100ppm
DG, DP	3%, 2° or less
Power consumption	2.2W (12V power supply)
Dimensions of transmitter	60mm × 45mm × 90mm
Weight (including antenna)	300g
〔Antenna〕	
Bandwidth	200MHz (VSWR < 2.0)
Gain	13dBi
Half-power beamwidth	29°
Dimensions	90mm × 90mm

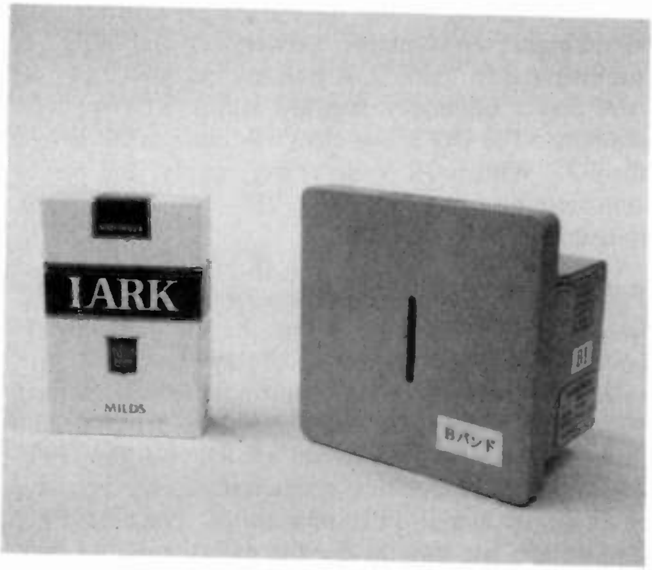

FIGURE 8: External appearance of practical FPU transmitter.

EXAMINATION OF A PROGRESSIVE COMPONENT TV SYSTEM WITH AN EYE TO MEDIA CONVERSION

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Tokyo, Japan

Abstract

Discussions are currently being conducted in Japan by the Telecommunication Technical Council and the Broadcasting Technology Association about how to meet the 1995 target for the introduction of the second-generation EDTV. Second-generation EDTV is a broadcasting system that provides high image resolution broadcasting with a 16:9 aspect ratio, yet maintains compatibility with the current NTSC system. For this purpose, it is desirable for broadcast station facilities to adopt a 525 progressive scanning system. What is needed is not only the incorporation of interlacing information into CCIR Recommendation 601 (commonly known as the 422), which is the international standard for digital component systems, but a new standard for progressive systems.

We report here on our study of a signal format in which a signal for providing progressive scanning information is added to the conventional 422 signal using the same development style. This system makes it possible to construct a system for not only EDTV, but also for systems other new media that conform to Japan's MUSE-HDTV, North America's ATV, Europe's PALplus, and other formats that interchange programs while maintaining constant picture quality.

1. Introduction

Trial broadcasting of MUSE-HDTV by satellite has already begun in Japan since 1991. In addition, digital broadcasting by satellite and cable has entered the experimental stage.

On the other hand, while NTSC is used in terrestrial broadcasting, large-screen household receivers have become popular, and problems such as interlacing flicker, and cross-color interference and cross-luminance interference caused by Y-C separation error are now becoming apparent.

Against this background, general TV viewers have come to expect high picture quality even from terrestrial broadcasting. There has been vigorous discussion in the Telecommunication Technical Council and the Broadcasting Technology Association on achieving the 1995 target for practical implementation of the terrestrial wide-screen, high-picture-quality second-generation EDTV. Improvement of picture quality through conversion of broadcasting station facilities to high-quality progressive, component, and digital systems is under study.

The current CCIR Recommendation 601, however, provides no standard suitable for progressive scanning. If we use 525 progressive signals, current operation requires [4, 4, 4]x2 or [4. 2, 2]x2 formats, which is to say 888 or 844, and a technique must be used to separate the two interlaced fields for each scan line. We, therefore, studied a component system in which a progressive-providing signal, +X, is added to the 422 format of the 525 interlaced signal.

2. 422 + X Component Format

As mentioned above, the current international standard for digital components is CCIR Recommendation 601. In that recommendation, the most frequently applied standard is 422 (the studio standard), represented by the D1-VCR. We have studied the 422+X format, an extension of this standard, in which a supplementary progressive-providing signal is added to the 422 [Y, R-Y, B-Y] format. This signal conveys the luminance information.

By adding to the interlace signal within the field, the luminance information serves as the progressive-providing signal that can carry the progressive signal. This new progressive signal carries twice the information of the interlaced signal however, so it is not compatible with conventional broadcasting as-is, and therefore making the system disadvantageous.

Therefore we studied a signal format (422+X) in which the progressive-providing signal of luminance is added to the interlace signal and the progressive signal is recovered. Since this signal format uses the ordinary fundamental interlace signal format, it is possible to handle current VCRs, switchers, and other such equipment like conventional ones. Thus it becomes possible to construct a low-cost system.

One 422+X signal generation method first separates the two interlace fields for each scan line ([4,2,2]x2). (See Fig. 1.) The information in the No.1 interlace field is used just as it is (CCIR Recommendation 601 422 studio standard). Thus, except for the progressive-providing signal, the signal format is completely the same as conventional 422, which makes it possible for 525 progressive systems to coexist with 525 interlace systems and to shift to them step by step. The luminance signal of the first interlace field is the 4 of 422, that is, sampling frequency is 13.5MHz x 8bits or more. The interlace No.2 field is used for luminance signal only; there is no color signal at all. Accordingly, although there is only half of the color signal in the vertical direction, the luminance signal is completely progressively recoverable.

Other than the method described above, we also investigated a new type of camera for use as the signal source. Using a CCD with a new driving method, this camera can simultaneously output two interlace signals. This camera makes it possible to construct a system of low overall cost.

To carry the +X information, we investigated the +4 and +2 formats, as well as a signal format that has only half of the color information of interlace No.1. These signal formats are described below.

2.1 422 + 4

In the case of 422+4, the luminance information is acquired by progressive scanning in the horizontal direction at about 12 MHz (equivalent to 6MHz interlaced scanning). Accordingly, it is possible to provide full-band luminance information in the both horizontal and vertical direction with 525 progressive scanning. Concerning color, however, the resulting resolution is only 3 MHz horizontally and 120cph vertically.

Previous subjective quality assessments have shown that 2MHz horizontal resolution and 100cph vertical resolution provides satisfactory picture quality.[1] Thus we conclude that 422+4 information is sufficient for 525 progressive display quality.

A three-dimensional spectrum for the 422+4 luminance information is shown in Fig. 2. It is desirable that the progressive-providing supplementary signal "4" is transmitted and recorded in the form of the luminance information of interlace No.2 as-is.

2.2 422 + 2

The band allocated to the progressive-providing supplementary signal in 422+2 is "2" (luminance information with a sampling frequency of at least 13.5/2MHz x 8bits or equivalent). The bandwidth allocated to the luminance of the No.1 interlace part is the 4 in 422.

In this case, the following three points were considered regarding the progressive-providing supplementary signal "2".

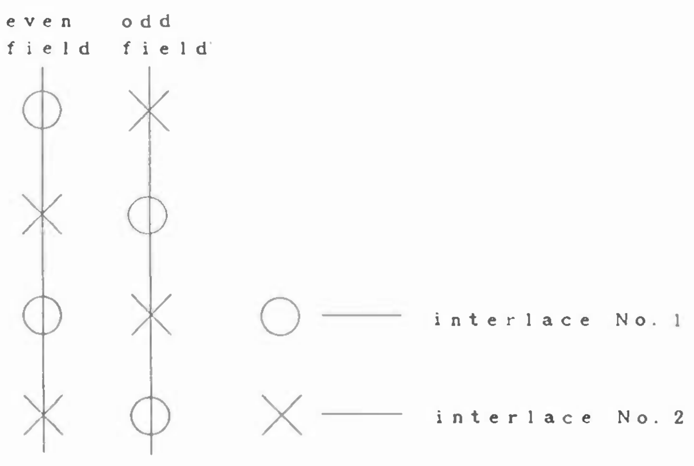


Fig. 1 [4, 2, 2]×2

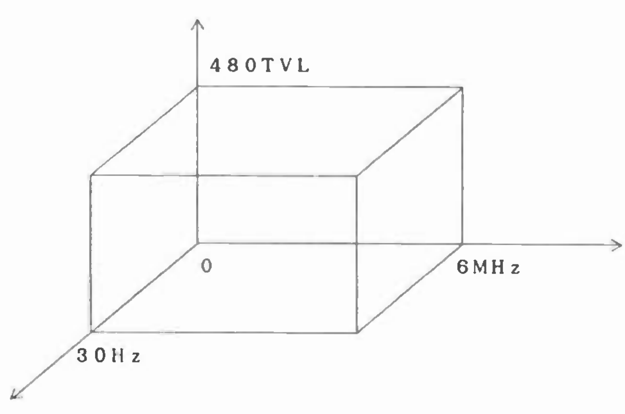


Fig. 2 3-dimensional spectrum of luminance for 422+4

1) The line difference (LD) signal

This LD signal is one type of supplementary progressive-providing signal. It can be expressed by the following equation.

LD = b - (a+c)/2 (See Fig. 3.)

It is equivalent to the 3-tap filter. The horizontal band of the supplementary progressive-providing signal is restricted to half-band, so the extremely high frequency component cannot be completely regenerated. Thus, it is necessary to filter this extremely high frequency component in advance so as to prevent interference caused by the aliasing.

It is a merit to use LD signal, the technique used in the second-generation EDTV, that sharp vertical high frequency component can be taken out because of using vertical 3-tap filter.

2) The 2-line difference signal (ΔY)

This ΔY signal is one type of supplementary progressive-providing signal. It can be expressed by the following equation.

Y= b - a (See Fig, 3.)

In the same way as for the ΔY signal, the extremely high frequency component cannot be completely regenerated. Thus, in this case also, it is necessary to remove this extremely high frequency component in advance so as to prevent interference caused by aliasing. A three-dimensional band diagram for when ΔY and LD are used is shown in Fig. 4.

It is a merit to use ΔY signal that we can make the hardware simple using vertical 2-tap filter.

3) The field difference (FD) signal

This FD signal is one type of supplementary progressive-providing signal. It can be expressed by the following equation.

FD = a - a' (See Fig. 3.)

A three-dimensional band diagram for when FD is used is shown in Fig. 5.

In this case, when the field difference is 0, that is, for the case of still pictures, it is a merit to use FD signal that the entire band 844 can be completely regenerated. For moving pictures, on the other hand, the horizontal extremely high-frequency component cannot be regenerated because of the band restriction. Therefore, it is necessary to remove this extremely high frequency component in advance so as to prevent aliasing in the same way as for the ΔY and LD signals.

In any of the three cases described above, the progressive luminance information is only half that of the 422+4 case. Nevertheless, even compared with the New NTSC which is proposed for the second-generation EDTV, the lost information is almost no difference between 422+4 and 422+2 in general pictures. Accordingly, we consider that 422+2 has potential for use in the programs of the second-generation EDTV.

2.3 411 + 2

For 411+2, the band allocated to the progressive-providing supplementary signal is the same

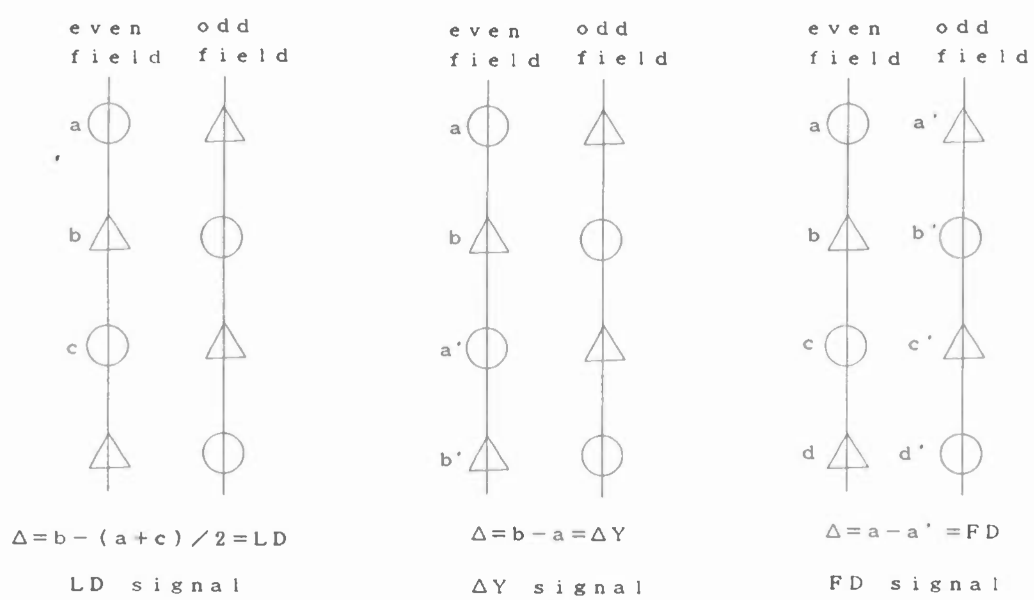


Fig. 3. LD, ΔY and FD signals

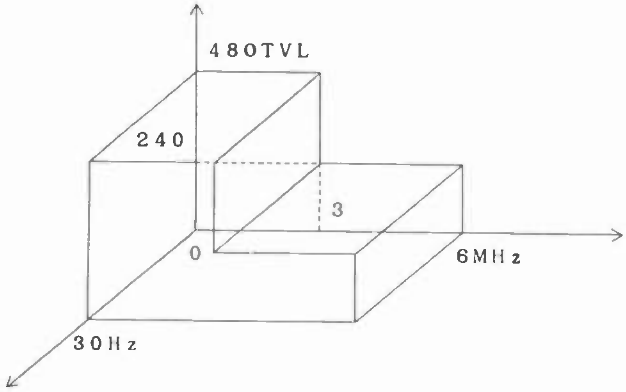


Fig. 4 3-dimensional spectrum of luminance for 422+2 (ΔY, LD)

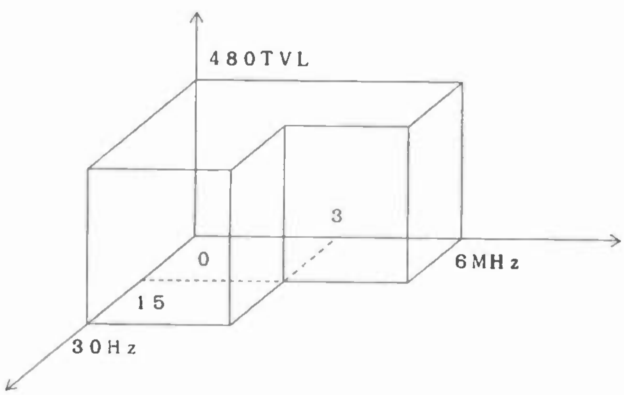


Fig. 5 3-dimensional spectrum of luminance for 422+2 (FD)

as for 422+2. Thus, the luminance frequency band that can be displayed is the same.

For the color difference information, on the other hand, while only 1.5 MHz can be reserved for the horizontal direction, a band of about this size is satisfactory because the amount of color information that can be transmitted by NTSC is I = 1.5 MHz and Q = 0.5 MHz. Accordingly, it is possible that simple cameras such as the ENG camera can be used.

3. Example of a System Using 422+X

An example of a system using 422+X is shown in Fig. 6.

I) Camera

The New NTSC being proposed by NTV as the second-generation EDTV uses the following three types of camera.

- HDTV cameras
- 525 progressive scanning cameras
- 525 interlace scanning cameras (2-line read-out CCD cameras)

The HDTV cameras and the conventional 525 progressive cameras are more expensive than the 525 interlaced scanning cameras. Thus, we studied a 525 progressive camera which uses a 2-line read-out CCD. By improving on the drawback of the 525 interlaced scanning camera, this camera records 525 progressive information. (See Fig. 7.)

Although CCD's normally have 525 lines of information, current NTSC cameras use 525 interlacing and mix each pair of lines to achieve high sensitivity and to prevent aliasing.

In our proposed method the 2-line mixing is not done, and the odd and even lines are read out separately. The information for each even and odd line is the same speed as the 525 interlace information, and as shown in Fig. 7., by mixing them, 525-progressive information is reproduced. The vertical information is therefore 240cph. The design of this 2-line readout camera is similar to that of existing interlaced cameras, so the price should not be much different from conventional 525 interlace cameras.

II) Switcher

A 422+4 switcher is currently available. However, the +4 part of this switcher is used for the key signal, so it has a possibility to use it as the supplementary progressive-providing information. We can use this switcher in the case of 422+4 and 422+2.

If we convert into the 411+2 signal format, there is a possibility that the existing 422 switcher can also be used.

III) VCR

The system that can currently be used with VCRs uses two D1-VCRs. One D1-VCR can record 216Mbps information. So, this system makes it possible to record the signal of 432Mbps. However, while systems that must synchronize the input of two VCRs are applicable for experimental use, they are not intended for use at actual broadcast sites. A practical system must be capable of recording 525 interlaced data together with the supplementary progressive-providing signal with a single camera.

VCRs for 422+4, 422+2 and so on are now being studied. In the case of 422+4, the VCR needs 324Mbps bit rate and in the case of 422+2, it needs 270Mbps bit rate. For 411+2, a 422 D1-VCR can be used as-is by adding an external signal switch, in the same way as described above for the 422 216Mbps switcher.

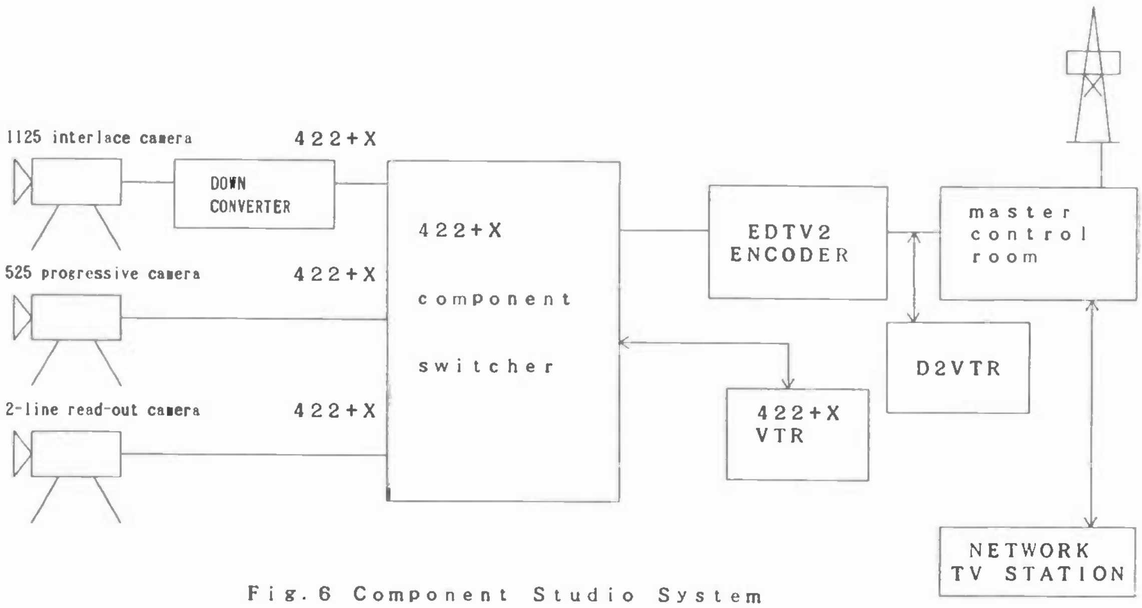


Fig.6 Component Studio System

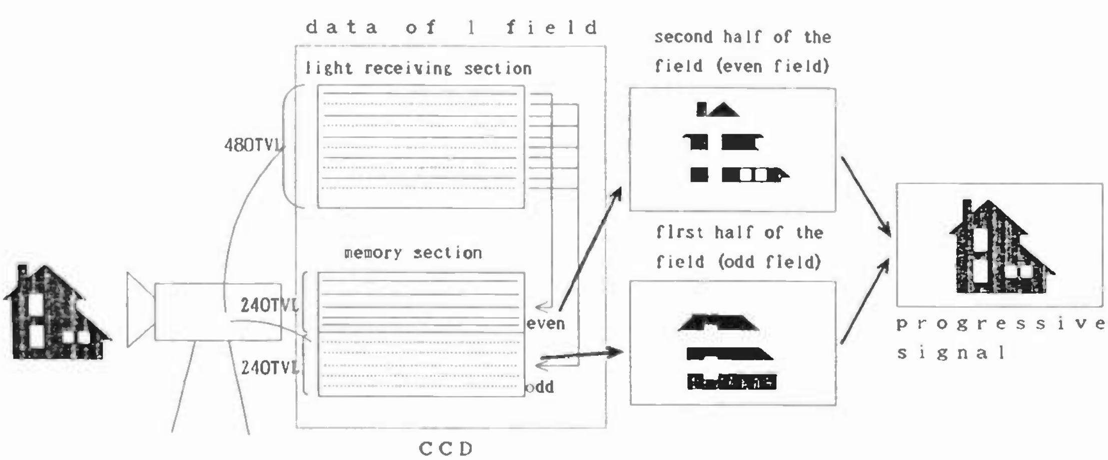


Fig. 7 2-line read-out CCD

4. Development Potential of this System

As described above, the 422+X system can provide a low-cost, high-resolution picture for the second-generation EDTV. This system also has potential for flexible application with respect to other media. (See Fig. 8.)[2] As an example, we did a software simulation of conversion to the 1125 interlace format.

The results of subjective quality assessment for pictures converted from a 422+4 signal to HDTV and then further subjected to MUSE conversion, and original HDTV pictures subjected to MUSE conversion. For 422+4, there is no perceived degradation in the picture quality.

For 422+2 and 411+2 however, as expected, while there was no perceptible difference for moving pictures, there was a rather significant difference in the still mode.

Judging from these results, we can conclude that 422+4 can provide high quality conversion to MUSE-HDTV system rather than conventional NTSC composite system. The probability is high that this system can also be used with formats other than MUSE-HDTV, such as the North American ATV and the European PALplus.

5. Conclusion

We have described the present state of 422+X systems, issues concerning them, and their potential for further development. In Japan, experimental satellite broadcasting in the MUSE-HDTV format has already begun. High picture resolution media for wide-aspect formats will be developed in the future by means of digital broadcasting by satellite and CATV. However, in Japan, a direct transition from the current NTSC broadcast format to these new broadcasting methods is not possible. We have attempted to achieve a second-generation EDTV system that extends NTSC to a wider aspect and higher picture quality. If this system can be applied for future new media broadcasting, too, it will prove to be exceptionally useful.

Thus, if the interlace component digital 422+X systems are constructed with the second-generation EDTV ones, it should be possible to provide good flexible and low-cost preparation for the coming wide-aspect, high-resolution, multimedia era. We hope that this report will contribute to the early realization of high-quality, wide screen broadcasting.

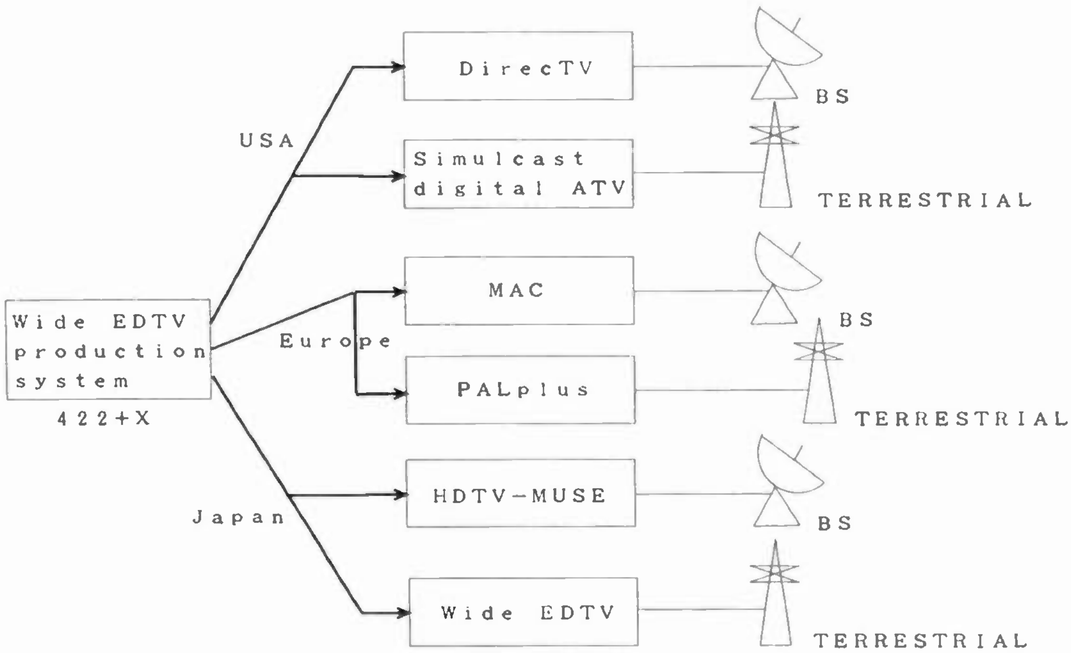


Fig. 8 Program exchange among USA, Europe and Japan

6. References

1) Yusuke Niitsu, etal., "Evaluation Experiments of Wide Aspect TV pictures (in Japanese)", 1990NAB Symposium of Broadcast Technology, Nov., 1990.

2) Kiyotake Fukui, "EDTV and ATV, Trends in the U.S. and European Countries", 1992 ITE Annual Convention, July, 1992.

MANAGING IN BROADCAST ENGINEERING

Monday, April 19, 1993

Moderator:
Margaret Bryant, WMAQ-AM, Chicago, Illinois

***MANAGING BY THE NUMBERS: AN EXAMINATION OF THE CHANGING IMPACT OF ENGINEERS ON BROADCASTING MANAGEMENT**
Brad Dick
Broadcast Engineering Magazine
Kansas City, Missouri

***IMPROVING THE PERCEPTION OF YOUR VALUE TO THE STATION**
Rip Riordan
KSAS-TV
Witchita, Kansas

***A FULL-TIME ENGINEER? OF COURSE I HAVE ONE!**
Richard Novik
WKIP-AM
Poughkeepsie, New York

THE CHANGING ROLE OF THE TELEVISION ENGINEER
George J. Csahanin
KXAS-TV
Fort Worth, Texas

SETTING GOALS AND MEETING THEM: TIPS FOR MANAGING YOUR BROADCAST ENGINEERING CAREER
Jerry E. Brown
WTRG-FM
Raleigh, North Carolina

***GETTING ON THE TEAM: MAKING THE ENGINEER A FULL PARTNER IN STATION MANAGEMENT**
Tom McGinley
Cook Inlet Radio Partners
Greenbelt, Maryland

COST EFFECTIVE ENGINEERING
Matt Sanderford
MARSAND, INC.
Fort Worth, Texas

***BROADCAST ENGINEERING MAINTENANCE MANAGEMENT**
Joseph Mahedy
Computer Assisted Technologies
New York, New York

*Paper not available at the time of publication.

THE CHANGING ROLE OF THE TELEVISION ENGINEER

George J. Csahanin
KXAS-TV
Fort Worth, Texas

Here is one Director of Engineering's way to seed idea growth. His goal is to adapt methods used in communicating with other departments in a TV station. The increased competitive pressures we feel today make some attitude change necessary, departing from older traditional mindsets.

Look at any article in Broadcasting, the magazine for and about our business. In a typical issue there were 44 articles and features. Of them only 7 were concerned with technology, mostly vague references to what some new techno-toy can do for revenues, and finally, three articles mentioned,or were about Rush Limbaugh. The point is that what we do... That is the broadcast engineer... Does not find its way into the premiere journal of this industry.

Now, you may feel like that's fine, "if they don't know I'm here in the basement, I'm safe." That couldn't be more wrong. If you get a paycheck, someone knows that you do. And if your company works like most the existence of every employee is questioned every year, at budget time. And every now and then one of our fold get zapped. Hardly anyone knew he existed or what he contributed, and a well maintained plant can go for some time before lack of preventative maintenance becomes noticeable.

We all know how much **we** contribute daily, but does anyone else have a clue as to what you do? People like the General Manager perhaps?

"What they might know about you" falls into a couple of categories. One is a memory of the last time you annoyed them. Another is the image of the last time you saved their tail. Hopefully the latter out-number the former. If not, there could be trouble.

Too many of us go to work, do a good job, make a few mistakes, have some victories, and go home. We as a group do not communicate our victories well. Others take care of communicating our defeats, (bad news sells).

I can tell you that better communications with your supervisors makes your continued employment possible. And that is what I'm getting to. I want to state what may be obvious, but needs saying. It's an old theme, but for lack of better words:

"Justify Your Existence"

I hate the sound of that because it has become a cliche, but it will have to do.

Now I don't mean that you should do 15 minutes of show and tell to the boss of what you did today. I'm talking about more subtle ways. Ones that aren't very direct, but very effective. I have a couple of favorite methods which have proven with time to be easy to implement-and are very effective. My idea here is to seed ideas into others minds to help improve their professional image.

Let's start with my #1 aid to information flow... The equipment trouble report - here's an example:
(see "sample wrong way)

KXAS-TV
Maintenance Report/Request

____
(Equipment Serviced)

Position ______________ S.N. # ________ Date ________

Emergency ( ) Non-Routine ( ) Routine ( )

Requested By ______________ Date Requested ________

Date/Time Removed from Service ______________

Date/Time Returned to Service ______________

Assignment and Location ______________

Complete Description of Problem: (Include tapes if video or audio problems are apparent on recording)

MAINTENENCE REPLY: ______________

List of Parts Replaced
Stock ______________

______________ Stock Level ______

Sp. Requisition ______________

____
Engineer Performing Maintenance

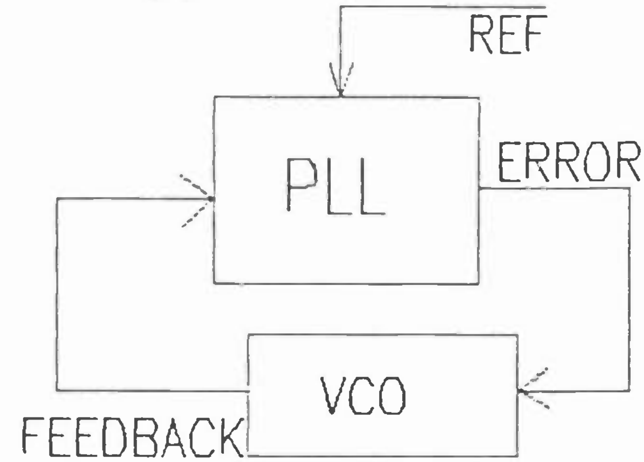
SAMPLE "WRONG" WAY

K???-TV ENGINEERING TROUBLE REPORT

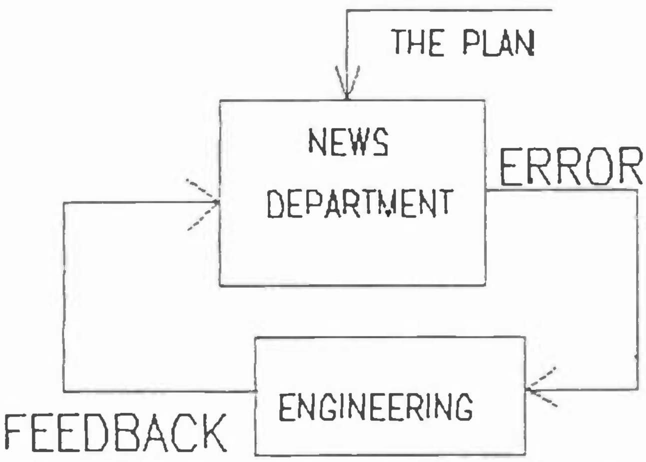
EQUIPMENT ______________

WHAT IS THE PROBLEM? ______________

It gives us all the info we need right? Wrong! I'll bet there are some of you who think this form is fine, but it lacks the aspect of being inter-active. At least not enough - there's no place for a reply. To close the loop.



We all know what a phase locked loop is. We also know what a news room is. Let's substitute some familiar words.



The important part of this analogy is the feedback part. If you don't give feedback after a requested engineering service has been performed, the "system" will unlock. I've found the simplest method of fixing this situation to be a more exotic trouble report form.

Shown here is the form currently in use at KXAS-TV. It is a five part NCR form. It requests facts that most news photographers probably, when first faced with, think are irrelevant. You tell them to trust you, and that, if they comply, they'll see a big change in shop work output...For them.

The truth is that, while your shop output may not actually have increased, they'll have written proof that will give them the impression that it has.

The written word lasts much longer than the spoken word. And in fact the shop output may grow because of the fact that on the input side of this form more information was available which can speed up the repair process.

And I guarantee that the newsroom people will let the news director know that things are being repaired faster. He'll be your ally and let the GM know that "stuff's happening." And that's good! I know it works - I've seen it happen in two stations so far.

This first method takes some effort as it requires journalists to stand still for more than a moment and fill out a form. Be patient. Eventually as many blanks as need to be filled in will be.

The second "trick" is to be visible. That means your whole department. The level of visibility takes conscious effort to do and manage. And it isn't designed to fool the news/production people into thinking you did something when in fact you have not. They need to know that you are their source for technology.

Visibility means do things... Time them to happen for exposure...**But not disruption.** They must know that you've acted on a problem. A part of this I've done for ten years now is to periodically change the bezel plastic on Sony BVU or BVW machines. The absolutely amazing part is how suddenly, magically, the machine has fewer "problems"-the ones where you spend several man-hours chasing a problem that may not have really existed in the first place. There wasn't an intentional effort to write up equipment for the fun of it in the past, but if a machine looks like it will be trouble it will be! The used car people found this out years ago. Shine it up and the customer will be less likely to think that it will be problematic. Unlike the used car people, never let a coat of wax take the place of quality repair. So, if the machine looks like it is in bad shape the customers will be expecting a problem. The same holds true for most equipment they use. And remember, the other departments of our stations are our **"customers"**

I'd like to turn to personal matters. The engineering manager needs to be a part of the station's team, even if the team doesn't want him or her on the court. You're in trouble if they don't want you at the game. Here are some other ways to make them want you around.

First, you need to be well groomed. Not to say a tux is in order, but your attire depends on the days actual tasks. I'd forgo the "nerd pack" pocket protector. If your appearance is disheveled others won't want to interface with you. And don't get me wrong - I don't mean to wear Hartz Shafner & Marx clothing to do antenna work but the idea should be obvious.

Periodically, make other department's personnel learn something about your world. At one station I took the GSM, and anyone else who would fit in the car to the transmitter site. The show and tell can be informative and you can point to what you've been trying to explain in staff meetings. For example, the klystrons you have had to purchase eating up the proceeds from ten super-bowl spots can be finally visualized. You can try to explain how they work, and show why you sometimes return from the transmitter looking like his Mercedes mechanic.

We often assume that the other department personnel have no interest in our world. This is not true. However in explaining the equipment you have to not get too in depth in the matter or you run the risk of losing their interest. If you think the GSM will not go for the road trip you're wrong. Plan it ahead, expect to reschedule it at least three times - it will happen and you will have an ally.

Also remember the two way street concept. If the GSM can make time to learn your world you can also learn his(or hers). It an important step that will give you more of the big picture.

What is the "Big Picture"? Have you ever sat in the office or shop and wondered why you have been asked to do a remote, and this remote seems like there is no reason to do it? There may be a very good reason that won't be obvious until you ask. But when you ask the sales people are going to key into your body language. If your position in opening the conversation is an adversarial one you may get an answer but not an explanation. You need to climb into the sales manager's world and try to understand his just as much as he needs to understand yours. One of the first things taught in Dale Carnegie training is that the best way to make a lasting relationship is to ask him or her about themselves-their favorite topic. So, ask the sales manager what is happening in his territory, ask the program director and news director how their lives are going. Join them for lunch sometime. Invite them! Their response will be good, just remember:Do not use this as a means to confront them about a problem. The problems will be solved easily if at first you build a bridge to the individual. That remote that makes no sense may be an attempt to roll out a new show, bring in a sponsor your station has been trying to get on the air for years or a number of reasons. The tendency is for engineers to complain about these things, but to do nothing about becoming enlightened.

I just mentioned Dale Carnegie. There are variations on the theme, and in fact variations within the offerings of that company but, I recommend very strongly that some kind of training like this be done by any engineering manager. In fact it would be good in general for most technicians to do it as well. I did and thought it was a lot of bunk. Then in later years I found myself saying things that I knew I would not have said the same way before. Its the old story of the glass being **half empty** or **half full**. The president of one of the leading broadcasting companies has a saying when things are not going well, which is "take the lemons and make lemonade". Think of the positive imagery in that statement. Nobody makes it to the top, or even moves closer , without a good positive "can do" attitude. Some of these people may be very demanding, but there are common themes. Ever notice in movies how the company CEO always has a certain look and attitude? It is stereotyped, indeed, but the meaning is there that the people that make it have certain qualities. These qualities can be acquired, if you want them.

Some people are born leaders. some need to be shown the way to lead. Some people, frankly, want to follow. That's ok. The world can't all be Fortune 500 CEO types people. However, if you are a broadcast technician or technical manager you have made a choice to be where you are today, probably more so than any other person at your station.

So you might as well excel at **it**. But "**it**" covers more territory than in the past.

Have you seen any dinosaurs lately? I bet you have. The guy at the station, maybe working for you, who is just convinced that anyone under 40 is some inexperienced kid who can't possibly know how to do tv(or radio). He knows this very well because he's been doing it for 40 years. He won't listen to the fact the he's been doing it a bit less efficient for the last twenty of those years. You hate it when this person appears at your office door because you just know it is to report on his latest "battle" with those kids in the news department. What do you do with him? I'll tell you. Feed him positive thoughts and ideas. You can slowly change him, but it is a great challenge. You may not have the time to devote to this goal, but it is worth a try because he does have a good deal of experience to cull from. But you have to separate the voice of experience from the grousing of someone who's had it all wrong for twenty years and doesn't realize it.

This prescription for survival is meant by me to reinforce ideas that you, the creative engineering manager, have already had. Your presence here in this room tells me that you have made it past the first hurdle-recognizing that broadcast engineering management has changed. Plenty of the change has been as a result of the change in the technology we deal with. It pained me see a board swap in a still store recently rather than a technician finding the bad IC and changing it. But this is how we do some things in the 90's. Shown here is that our technology isn't all that must adapt to the 1990's.

Your boss has to kick and scratch for more revenues to make the quarter end in the black than 20 years ago. He's got to work toward the "plan." You need to work with him. The best way to do so is to have a realistic business plan in place for both expenses and capital investment in the facility. Remember, you are not buying those new toys to have and play with. They are purchased because your company is investing in the property. It makes the station as a whole more valuable. The expense side of things must be managed tightly. You must be able to plan one year ahead. Plan on the year's regular remotes. Budget for them. The real black and white evidence of a **plan** is the written budget. Perhaps in future NAB shows we can have a seminar for Engineering people on how the financial end of this business works. A budget working according to plan is a thing of beauty. It should not be derived by taking last years numbers and adding 5%. This doesn't cause anyone to think about what is really going to be done. If all expenses are over and above the budget then the GM will have a tough time making it work, and if that tube replacement you did put the month, quarter, or year in the red you'll be the bad guy, unless it was part of the **plan**. Remember what I said in the beginning. You don't want the boss to only associate your name with the last time you did something the got him upset.

It also couldn't hurt to have the news director tell you, off the record perhaps, what his plans are for special coverage in the next budget year. That also helps you to plan.

Last October I said to the Texas Association of Broadcasters' managers group that it is very important to get some clerical assistance for the engineer. It is something that many General Managers have difficulty understanding, but there is more paper flow in engineering than there is traditionally thought to be. I've had secretaries that were real sharks. One was so good, and learned so much about the stations operations that I promoted her to Operations Manager.

To conclude this I will admit that most of my theories are based on 13 years of management and several more as a good observer. The least likely place where you will need to work on your management skills in the future will be in dealing downward toward the "troops". You already, probably, have a good understanding with them. But if you want to survive in the business you need to be sharp at getting along with and understanding more than just the technical. Most people in my position have done just that. Those who have not changed with the times may be facing trying times.

I wanted to show here that a degree in management is not required to survive, just some common sense skills aimed toward adapting to the 90's.

SETTING GOALS AND MEETING THEM: TIPS FOR MANAGING YOUR BROADCAST ENGINEERING CAREER

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ABSTRACT

THE IMPORTANCE OF GOALS IS DISCUSSED, AS WELL AS A METHOD FOR SETTING GOALS. ADDITIONALLY, TIPS FOR BETTER TIME MANAGEMENT, WRITTEN AND ORAL COMMUNICATIONS ARE PROVIDED.

INTRODUCTION

BROADCAST ENGINEERING IN MY OPINION RATES RIGHT UP THERE WITH THE MAYTAG REPAIRMAN, AS ONE OF THE LONELIEST JOBS IN AMERICA. WHY IS THAT? IN THE CLIMATE OF DOWN SIZING, AND ALMOST COMPLETE TECHNICAL DEREGULATION, THE FIRST DEPARTMENT TO BE EVISCERATED OF PERSONNEL WAS ENGINEERING. NOW MOST RADIO STATIONS ONLY HAVE A CONTRACT ENGINEER; THE LARGER STATIONS, A ONE PERSON DEPARTMENT.

BECAUSE OF THIS, IT IS EXTREMELY DIFFICULT FOR ENGINEERS TO BECOME MOTIVATED, MUCH LESS REMAIN "UP". BUT I ARGUE IT IS ESSENTIAL FOR ALL OF TO REGAIN THAT MOTIVATION. RATHER THAN SPEND THE NEXT 30 MINUTES BEMOANING OUR PLIGHT, LET'S MOVE ON AND DEVELOP A PLAN TO SET ATTAINABLE GOALS, AND BE THE BEST WE CAN BE, THUS PROVIDING A SECURE FUTURE FOR OURSELVES, OUR FAMILIES, AND IN THE PROCESS ACHIEVING JOB SATISFACTION.

FIRST LET'S DEAL WITH GOALS. GOALS ARE A ROAD MAP TO ASSIST YOU IN FINDING YOUR DESTINATION. FOR THE MOST PART MOST OF US NEVER SET OUT FOR A CITY WHERE WE HAVE NEVER BEEN WITHOUT A MAP AND/OR A SET OF DIRECTIONS. BUT WHEN IT COMES TO THE MOST IMPORTANT TRIP WE WILL EVER MAKE, LIFE, WE MUDDLE THROUGH WITHOUT A THOUGHT OF WHERE WE ARE GOING, OR HOW TO GET THERE. IF YOU REMEMBER EARLIER IN THIS TALK, I MENTIONED GOALS, BUT AS YOU RECALL I SAID ATTAINABLE GOALS. AND THERE'S THE KEY TO SUCCESS!

IN DEALING WITH SUCCESS, KEEP IN MIND SUCCESS IS NOT MEASURED BY YOUR W-2, THE SIZE OF YOUR HOUSE OR THE MAKE OF AN AUTOMOBILE. SUCCESS IS AS INDIVIDUAL AS YOUR FINGERPRINTS. OF COURSE WE AS A SOCIETY HAVE PLACED VALUES ON MATERIAL ITEMS, AND AS A SOCIETY ATTACHED THE MEANING OF SUCCESS TO THOSE ITEMS. BUT IS IT REALLY?

GOALS

HERE ARE MY THOUGHTS ON DEVELOPING A PERSONAL PLAN OF SUCCESS:

1. DETERMINE WHERE YOU ARE IN YOUR LIFE PRESENTLY. DO NOT SPEND TIME ON HOW YOU GOT THERE, IT DOESN'T MATTER.

YOU MUST ALWAYS LIVE IN THE PRESENT, REMEMBERING FROM WHERE YOU CAME, BUT NOT DWELLING ON IT.

2. DECIDE WHAT YOU WANT FROM LIFE: MONEY, MORE MONEY, CARS, A FAMILY, A BETTER JOB, HAPPINESS IN YOUR PRESENT JOB?

NOW I MEAN REALLY SPEND TIME THINKING ABOUT THESE THINGS. ONCE YOU'VE DECIDED ON THESE THINGS, WRITE THEM DOWN. THESE NOW BECOME YOUR DESTINATION.

3. IN ORDER TO REACH YOUR DESTINATION YOU MUST NOW DEVELOP A SET OF GOALS, DIRECTIONS, A ROAD MAP IF YOU WILL, ON HOW TO GET THERE. THIS IS VERY COMPLEX AND REQUIRES A GREAT DEAL OF THOUGHT AND TIME. WHEN SETTING GOALS, IT IS VERY IMPORTANT THAT EACH GOAL MEET TWO (2) CRITERIA;

A. PROGRESS TOWARD THAT GOAL MUST BE MEASURABLE

B. THE GOAL MUST BE ATTAINABLE.

WHY? FOR YOUR OWN MENTAL HEALTH!

4. IN DECIDING YOUR LIFE'S MAJOR GOALS I HAVE FOUND IT HELPFUL TO DO SOME REVERSE ENGINEERING. START WITH A DESTINATION, THEN DEVELOP A PLAN TO GET THERE. I HAVE FOUND THE TEN YEAR PLAN TO BE THE BEST. DECIDE WHERE IN LIFE YOUR REALLY WANT TO BE IN TEN YEARS. IT'S OK IF IT SOUNDS OUTLANDISH. WRITE IT DOWN. NEXT IN ORDER TO ACHIEVE THIS GOAL, WHERE WOULD YOU NEED TO BE IN FIVE (5) YEARS: TWO (2) YEARS, AND IN ONE (1) YEAR? MAKE A LIST OF THINGS THAT YOU CAN DO DAILY TO ACHIEVE YOUR FINAL GOAL.

5. ONCE YOU HAVE WRITTEN DOWN THE GOAL(S) AND THE 10 YEAR PLAN, PLACE THEM IN A PROMINENT PLACE; THE FRONT OF YOUR DAY PLANNER, YOUR DESK, ETC. EVERYDAY WHEN YOUR ARE PREPARING YOUR "MUST BE DONE" LIST, WRITE DOWN ONE OF THOSE DAILY ITEMS. CONTINUE TO DO THIS AND YOU WILL REACH YOUR GOALS.

CHARLES GARFIELD IN HIS BOOK **PEAK PERFORMERS** LIST SIX ATTRIBUTES OF THE PEAK PERFORMER. THEY ARE:

*MISSIONS THAT MOTIVATE
*RESULTS IN REAL TIME
*SELF-MANAGEMENT THROUGH SELF MASTERY
*TEAM BUILDING/TEAM PLAYING
*COURSE CORRECTION
*CHANGE MANAGEMENT

I BRING THESE TO YOUR ATTENTION, BECAUSE OF THE LAST TWO ITEMS, COURSE CORRECTION, AND CHANGE MANAGEMENT. WHEN YOU SET A GOAL, YOU MAY REACH IT SOONER THAN EXPECTED. DON'T STOP HERE SET NEW AND HIGHER GOALS! ALSO SOMETHING UNFORESEEN COULD DELAY YOU REACHING A GOAL, SO WHEN SETTING A GOAL, YOU MUST BUILD IN ALLOWANCES FOR COURSE CORRECTIONS. GAINING THE ABILITY TO MANAGE CHANGE IS A KEY ELEMENT UNFORESEEN COULD DELAY YOU REACHING A GOAL, SO WHEN SETTING A GOAL, YOU MUST BUILD IN ALLOWANCES FOR COURSE CORRECTIONS. GAINING THE ABILITY TO MANAGE CHANGE IS A KEY ELEMENT CHANGE. ENGINEERS AS A GROUP DO NOT HANDLE CHANGE VERY WELL. WE ALL GET INTO OUR COMFORT ZONES, AND RESIST ANY TYPE OF CHANGE. PLEASE IF YOU DON'T REMEMBER ANY THING ELSE I SAY, LEARN TO ADAPT YOURSELF TO THE SITUATION, SPEND A LITTLE TIME ANALYZING IT, BEFORE YOU MAKE A JUDGMENT.

APPEARANCE

PROBABLY ONE OF THE BIGGEST PROBLEM AREAS FOR BROADCAST ENGINEERS IS GAINING THE RESPECT THEY DESERVE. I ARGUE A PARTIAL CAUSE FOR THIS CAN BE LINKED DIRECTLY TO APPEARANCE. IN 1986 BROADCASTING MAGAZINE PUBLISHED AN ARTICLE ENTITLED "A PROFESSIONAL LOOKS THE PART." IF YOU WERE TO GO TO MY OFFICE TODAY, YOU WILL SEE THE ARTICLE FRAMED AND IN A PROMINENT PLACE. THIS ARTICLE WAS RIGHT ON TARGET. IT POINTS OUT THERE ARE SEVEN DECISIONS PEOPLE MAKE BASED SOLELY ON YOUR APPEARANCE:

1.) INCOME AND ECONOMIC LEVEL
2.) EDUCATIONAL BACKGROUND
3.) SOCIAL POSITION
4.) LEVEL OF SOPHISTICATION
5.) DEGREE OF SUCCESS
6.) MORAL CHARACTER
7.) TRUSTWORTHINESS

AS THE COMMERCIAL SAYS, "YOU NEVER GET A SECOND CHANCE TO MAKE A FIRST IMPRESSION."

WHAT DOES YOUR EVERYDAY APPEARANCE SAY ABOUT YOU AND YOUR POSITION ON THE ABOVE SEVEN ITEMS? RIGHT OR WRONG, THE BUSINESS WORLD PLACES AN INORDINATE AMOUNT OF IMPORTANCE ON HOW ONE LOOKS.

TO GAIN RESPECT, YOU MUST FIRST LOOK AND ACT LIKE YOU DESERVE RESPECT!

REGARDLESS OF THE SITUATION, YOU MUST ALWAYS PAY ATTENTION TO HOW YOU LOOK. I RECOMMEND STRIKING A BALANCE; NEATLY ATTIRED, YET FUNCTIONAL. IF YOU DEVELOP GOOD DRESS HABITS, YOU WILL ALWAYS BE DRESSED FOR ANY OCCASION. YOU NEVER KNOW WHO MAY DROP BY A BROADCAST STATION.

IN BUSINESS AND ENGINEERING, YOU SHOULD NEVER PRESENT A PROBLEM WITHOUT A SOLUTION. SO HOW DO YOU STRIKE A BALANCE?

FIRST, TAKE STOCK OF YOUR EXISTING WARDROBE. THEN SPEND SOME TIME LOOKING AROUND THE STATION. HOW ARE THE SALES PEOPLE DRESSED, THE SUPPORT STAFF, AND MANAGEMENT? ARE YOU DRESSED AS WELL, OR BETTER THAN THE MAJORITY OF THESE PEOPLE? IF NOT, YOU'VE GOT SOME WORK TO DO.

WHEN I WAS A CHIEF ENGINEER OF A RADIO STATION, I ALWAYS CARRIED A PAIR OF JEANS, SHIRT. AND SNEAKERS IN MY CAR. THAT WAYS, SHOULD A PROBLEM ARISE, I COULD THROW ON THESE CLOTHES AND NOT WORRY ABOUT MY GOOD CLOTHES.

YOU MUST DECIDE TO MAKE SEVERAL INVESTMENTS IN YOUR SELF AND YOUR CAREER. HERE ARE MY SUGGESTIONS FOR WARDROBE: I RECOMMEND ONE NICE DARK YEAR-ROUND WOOL BLEND SUIT. YOU NEED AT LEAST TWO COORDINATED TIES FOR THE SUIT. AS AN ENGINEER, STAY WITH WHITE OR BLUE OXFORD SHIRTS. IF YOU ARE A PRACTICING ENGINEER, I SUGGEST A 60/40 BLEND SHIRT, THEY DO NOT WRINKLE AS MUCH. MAKE SURE YOU HAVE YOUR SHIRTS LAUNDERED. AND ONE OTHER NOTE, MAKE SURE YOU HAVE BLACK SHOES TO WEAR WITH THE SUIT. THE SUIT WILL PAY FOR ITSELF IN NO TIME IN ADDED RESPECT. YOU CAN WEAR FOR IMPORTANT STAFF MEETINGS, SEMINARS, CONFERENCES, ETC.

FOR EVERY DAY WEAR, PURCHASE THREE OR FOUR PAIRS OF SLACKS. IN COLORS SUCH AS NAVY BLUE, BLACK, KHAKI, AND GRAY. STAY WITH OXFORD SHIRTS. YOU CAN ALSO STRETCH YOUR CLOTHING BUDGET, BY PURCHASING COORDINATING TIES, AND ONE NAVY BLUE BLAZER.

NEXT, GROOMING IS ESSENTIAL. LONG HAIR, SHORT HAIR, OR NO HAIR, KEEP IT NEAT. ALWAYS KEEP TOOTH PASTE, MOUTH WASH, SOAP FOR YOUR HANDS, AND A COMB CLOSE BY. FINALLY, APPEARANCE ALSO FACTORS INTO YOUR OFFICE, AND SHOP AREAS. TOM PETERS WROTE IN "A SEARCH FOR EXCELLENCE", THAT TRAVELERS EQUATE THE MECHANICALLY SOUNDNESS OF AIRPLANES WITH THE CLEANLINESS OF THE TRAY TABLES.

SO IF YOU WERE TO APPLY THIS LOGIC TO BROADCAST ENGINEERING: THE MANAGER OF THE RADIO STATION EQUATES THE CONDITION OF THE EQUIPMENT WITH THE CONDITION OF YOUR OFFICE.

COMMUNICATIONS

AS WE CONTINUE ON OUR PATH OF MANAGEMENT DEVELOPMENT, LET'S DISCUSS COMMUNICATIONS. IT IS SO IMPORTANT THAT THERE BE AN UNDERSTANDING OF EXPECTATIONS, FROM PROGRAMMING, MANAGEMENT, AND ENGINEERING. WE DISCUSSED THE STIGMA ATTACHED TO THE ENGINEER BY MANAGEMENT EARLIER. IN OUR COMMUNICATIONS, EITHER VERBALLY OR WRITTEN, WE DO LITTLE, IF ANYTHING TO DISPEL THAT STIGMA. SO OFTEN I HEAR COMPLAINTS FROM STATION MANAGERS: THEY SIMPLY DO NOT UNDERSTAND WHAT THE ENGINEER IS TRYING TO TELL THEM. I HAVE MADE SOME GOOD MONEY ACTING AS THE INTERMEDIARY FOR MANAGEMENT AND THE LOCAL ENGINEER. SO HERE I'M GIVING YOU THE SECRET TO MY SUCCESS. MANAGERS DO NOT WANT THE SPECIFICS OF A PROBLEM. THEY WANT TO KNOW SOME BASIC THINGS: HOW LONG WILL IT TAKE TO FIX IT?, HOW MUCH?, AND WHAT CAN WE DO TO PREVENT IT FROM HAPPENING AGAIN?

THE SPEAKING OF "ENGINEERESE" MUST BE ADDRESSED. WE ALL DO IT!. THE KEY TO SUCCESSFUL COMMUNICATIONS IS WHEN ALL PARTIES INVOLVED UNDERSTAND EACH OTHER. SO OFTEN, ENGINEERS ASCRIBE TO THE OLD THEORY THAT IF YOU CAN'T DAZZLE THEM WITH BRILLIANCE, BAFFLE 'EM WITH (YOU KNOW THE WORD HERE.) WELL, LET ME TELL YOU, THAT IS THE QUICKEST WAY OUT THE DOOR.

REMEMBER, YOUR STATION MANAGER, OWNER, OR IMMEDIATE SUPERVISOR DOES NOT EXIST IN A VACUUM. IF HE THINKS YOU ARE NOT BEING STRAIGHT WITH HIM, HE WILL CALL SOMEONE ELSE TO DECIPHER IN PLAIN LANGUAGE WHAT YOU HAVE SAID.
WHEN TALKING WITH NON-TECHNICAL PERSONNEL, KEEP IT SIMPLE AND HONEST. EVEN IF IT INVOLVES SAYING THE MUCH DREADED PHRASE, "I DON'T KNOW."

HENRY FORD, IN DEFENDING HIMSELF AGAINST A CHARGE OF STUPIDITY, SAID " I MAY NOT KNOW, BUT I KNOW SOMEONE WHO DOES."

THE NEXT PET PEEVE OF MOST OWNERS AND STATION MANAGERS INVOLVES WRITTEN COMMUNICATIONS. AS WITH YOUR APPEARANCE, YOUR WRITING STYLE TELLS A LOT ABOUT YOU. FOR GOD'S SAKE, AND MORE IMPORTANTLY YOURS, TAKE THE TIME TO MAKE SURE EVERYTHING IS SPELLED CORRECTLY, AND THE GRAMMAR IS UNDERSTANDABLE.

A PERSONAL EXPERIENCE

Several years ago I was working as a Chief Engineer for a very structured company. As was everyone, I was trying to climb the career ladder. At this time there was no Director of Engineering, so I had decided this was the post for me. The station I was assigned had a problem with lightning strikes on the new tower site. This was the first time this company had built such a large tower at any of the radio properties.

The Controller of the company called upon me to take thirty days and prepare a written report on what other companies with similar tower installations were doing to prevent this type of damage. The report would be presented to the President, Vice-President of the Radio Group, Controller, and the other engineers.

I spent the next thirty days traveling the country side visiting similar installations, talking with other engineers and consultants about the problem. The day before the report was due, I was still dictating my findings and recommendations. The next morning I arranged for the secretary assigned to my department to transcribe the report, make copies, and assemble the final package. Everything was going as planned, until she became ill, and left. There I was, two hours before the meeting with nothing but three (3) mini-cassettes of my work. I hastily arranged for another secretary to transcribe the report. Finally, 30 minutes before show time it was complete. No time to read and check the report. It wasn't necessary. I had been dictating reports for years, and all the dictation was in proper form for the transcriptionist. The transcriptionist had misspelled the word lightning, consistently throughout the report (45 times). The meeting went great, the President of the company graded my report: A for content, D for grammar and spelling.

It was another two years before I became the Director of Engineering. I still have the graded version of that report, I guess I always will.

If you have a computer, then you have the financial wherewithal to purchase a spell checker, grammar checker, and thesaurus. If you don't have a computer, spend three dollars for a dictionary. Then use them.

REPORTS

WHEN YOU ARE WRITING A REPORT, MEMO, LETTER, OR REQUEST, GET TO THE POINT IN THE FIRST PARAGRAPH. DON'T DANCE AROUND FOR THE FIRST PAGE, THEN MAKE YOUR CASE, AND FINALLY MAKE YOUR POINT. HIT THEM UP FRONT, THEN PROVIDE SUPPORTING INFORMATION. ALL EXECUTIVES, MANAGERS, AND OWNERS ARE AS BUSY AS YOU ARE, AND THEIR TIME IS IMPORTANT TO THEM. THEY APPRECIATE ANYONE WHO IS SENSITIVE TO THIS.

WHEN YOU ARE REQUESTING A NEW PIECE OF GEAR, WHETHER IT'S FOR YOUR SHOP OR THE PRODUCTION ROOM, EXPLAIN IT IN TERMS OF RETURN ON INVESTMENT, R-O-I. FOR EXAMPLE: IF YOU NEED A NEW OSCILLOSCOPE, YOU CAN MAKE THE ARGUMENT THAT THE STATION WILL EXPERIENCE FASTER REPAIR TIMES FOR CRITICAL EQUIPMENT.

FINALLY, WHEN PREPARING REPORTS, PRESENTATIONS, OR PROPOSALS, SPEND A LITTLE TIME DRESSING THEM UP. YOU WOULD BE SURPRISED HOW FAR A 99 CENT REPORT COVER CAN GO FOR A PRESENTATION IN A STAFF MEETING. ALSO, MAKE SURE YOU HAVE COPIES FOR EACH OF THE KEY PEOPLE INVOLVED.

TIME MANAGEMENT

THE KEY TO ANYONE'S SUCCESS IS DIRECTLY PROPORTIONAL TO THEIR ABILITY TO ORGANIZE THEMSELVES. SUCCESSFUL PEOPLE ARE VERY AWARE OF THEIR TIME. THERE IS NO WAY THE LIMITED AMOUNT OF TIME I HAVE HERE TONIGHT TO PROVIDE YOU WITH A THE NECESSARY ITEMS TO HELP YOU MASTER TIME MANAGEMENT. SO WHAT I WOULD LIKE TO DO IS EXPOSE YOU TO WHAT HAS WORKED FOR ME.

THERE ARE SEVERAL GREAT TIME MANAGEMENT SYSTEMS, THE ONE I PREFER IS THE DAYTIMERS SYSTEM. MY COMPLAINT WITH DAYTIMERS, IS THE SYSTEM ITSELF IS NOT VERY USER FRIENDLY. YOU MUST SPEND SOMETIME LEARNING HOW TO EFFECTIVELY USE IT. I RECOMMEND READING DR. CHARLES HOBBS' BOOK TIME POWER, OR LISTEN TO THE AUDIO CASSETTES OF THE BOOK. I PREFER THE CASSETTES SINCE I SPEND A LOT OF TIME IN MY CAR.
RECORD KEEPING IS AN AREA I WOULD LIKE TO SPEND A FEW MINUTES ON. IF YOU ARE USING, OR BEGIN TO USE THE DAYTIMER SYSTEM, YOU WILL FIND IT AN EXCELLENT PLACE TO CENTRALLY LOCATE ALL YOUR VITAL INFORMATION.

AS ENGINEERS, WE TEND TO KEEP RECORDS ON EQUIPMENT, BUT WE ARE NOT VERY WELL WHEN DEALING WITH PROBLEMS OR PROJECTS. ALWAYS KEEP TRACK OF WHO YOU SPEAK WITH, THE DATE AND TIME YOU SPOKE WITH THEM, AND A BRIEF DESCRIPTION OF WHAT YOU SPOKE ABOUT. I BELIEVE THIS CAN SAVE YOU LOT'S OF HEADACHES IN THE FUTURE. I RECOMMEND USING A MICRO CASSETTE RECORDER. THEY ARE INEXPENSIVE, THE TAPES ARE CHEAP, AND IT'S EASY TO USE. ALL YOU NEED TO DO IS SPEAK INTO THE MIKE, RECORD THE DAY, DATE, TIME AND A BRIEF DESCRIPTION OF THE SUBJECT MATTER. YOU CAN THEN FILE THEM BY MONTH. SHOULD A PROBLEM ARISE, YOU CAN EASILY REFER BACK TO THEM.

A WORD OF CAUTION ON RECORD KEEPING. TREAT EVERYTHING YOU DO WITH RESPECT. THERE IS NOTHING QUITE LIKE HAVING YOUR PERSONAL FILES SUBPOENAED. I SUGGEST, NEVER WRITE OR RECORD PERSONAL OPINIONS ABOUT OTHERS.

FINALLY ON THE SUBJECT OF ORGANIZATIONAL SKILLS, SET UP YOUR OFFICE AND SHOP IN A MANNER WHERE YOU PAY ATTENTION TO THE MATTERS AT HAND. SO OFTEN, WE GET CAUGHT UP IN OUR OWN LITTLE WORLD. MAKE SURE YOUR "WORKING PROBLEM" LIST IS THE SAME AS THE MANAGER'S. IF YOUR STATION DOES NOT HAVE A REGULARLY SCHEDULED STAFF MEETING WHERE YOU DISCUSS PROBLEMS,

I SUGGEST APPROACHING THE MANAGER ABOUT SETTING UP A TIME FOR YOU, THE PROGRAM DIRECTOR, AND MANAGER TO MEET. THIS WILL ALLOW YOU TO FIND OUT WHAT THE OTHER PEOPLE IN THE STATION SEE AS A PROBLEM. PLUS NO LONGER WILL THEY SEE YOU AS SOME STRANGE, NOT HUMAN INDIVIDUAL WHO EATS IC'S FOR BREAKFAST. OPEN UP THAT LINE OF COMMUNICATION.

MISCELLANEOUS TIPS

NEXT LET'S TALK ABOUT SOME INEXPENSIVE WAYS TO KEEP UP WITH WHAT'S GOING ON IN THE WORLD OF ENGINEERING. TRADE JOURNALS, AND MANUFACTURER'S "CUT-SHEETS."

EVERY MONTH I SPEND A DAY GOING THROUGH ALL THE TRADE JOURNALS MY OFFICE RECEIVES. I SCAN THE ADS FOR INFORMATION ON NEW PRODUCTS I AM NOT FAMILIAR WITH. I FILL OUT THE "BINGO-CARDS" AND MAIL THEM IN. NEXT, WE HAVE A FILE ON EQUIPMENT LITERATURE. THERE WE KEEP CUT SHEETS ON PRODUCTS, FILED BY MANUFACTURER. EACH SHEET IS STAMPED WHEN IT'S RECEIVED WITH THE MONTH AND YEAR. THAT MAKES IT EASY TO CLEAN OUT THE FILES.

BACK TO THE JOURNALS. AFTER REVIEWING THE ADS, NEXT I CLIP OUT ARTICLES. THESE ARE THEN FILED BY SUBJECT, AND PLACED IN SEVERAL HUGE THREE RING BINDERS. HERE IS MY BRAIN TRUST. SHOULD A PROBLEM OR PROJECT ARISE THAT I OR A STAFF MEMBER IS NOT FAMILIAR, THEN WE CONSULT THE NOTEBOOKS. BELIEVE ME THIS SAVES TIME AND MONEY. YOU DO NOT HAVE TO REINVENT THE WHEEL EVERY TIME.

WE SPOKE EARLIER ABOUT INVESTING IN YOURSELF WITH CLOTHING. ANOTHER INVESTMENT IS YOUR PERSONAL REFERENCE LIBRARY. JOIN SOME BOOK CLUBS, BOTH TECHNICAL AND OTHERWISE. THIS WILL ALLOW YOU THE OPPORTUNITY TO BROADEN YOUR HORIZONS AND BECOME A RESOURCE YOURSELF.

OCCASIONALLY, I PRACTICE WHAT I PREACH AND SPEND SOME TIME ALONE DISCUSSING WITH MYSELF MY ACHIEVEMENTS, AND MORE IMPORTANTLY MY SHORTCOMINGS. DURING ONE OF THESE TIMES I DISCOVERED I HAD NO FISCAL SKILLS WHATSOEVER. YEAH, I WAS LIVING COMFORTABLE, BETTER THAN MY PARENTS, I HAVE A NICE HOUSE IN THE COUNTRY, MINI-VAN, WIFE, THREE KIDS, AND A DOG. BUT WHAT HAPPENS TO THIS IF I BECAME ILL AND COULD NOT WORK? WHAT IF I WERE TO DIE SUDDENLY? SCARY HUH?

IF YOU HAVEN'T GIVEN ANY THOUGHT ABOUT THESE MATTERS, THEY ARE SERIOUS AND NEED IMMEDIATE ATTENTION. SPEND SOME TIME WITH A FINANCIAL PLANNER, AND DEVELOP A PLAN.

JOB HUNTING

FINALLY LET'S TALK ABOUT JOBS, AND HOW TO GET THEM. ENCLOSED IN THE MATERIAL I AM LEAVING IS A COPY OF A PAMPHLET FROM A HEAD-HUNTING FIRM. REMEMBER IN SEARCH FOR AN ENGINEER THE SAME RULES APPLY TO THE ENGINEERING CANDIDATE THAT APPLY TO THE STATION MANAGER CANDIDATE, OR SALES POSITION CANDIDATE.

I SAY AGAIN "YOU NEVER GET A SECOND CHANCE TO MAKE A FIRST IMPRESSION."

RESUMES SHOULD BE PREPARED IN A MANNER LISTING WORK EXPERIENCE, INCLUDING THE TYPE OF EQUIPMENT YOU ARE EXPERIENCED WITH, YOUR EDUCATIONAL HISTORY, AND YOUR PHILOSOPHY.

TAKE CHARGE OF THE JOB SEARCH! INTERVIEW PROSPECTIVE EMPLOYERS AS THEY INTERVIEW YOU. NEGOTIATE FOR THE BEST POSSIBLE DEAL, NOT JUST THE SALARY...HEALTH, LIFE INSURANCE, RETIREMENT, ETC.IT'S UP TO YOU..............

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COST EFFECTIVE ENGINEERING SURVIVING BOTTOM LINE MANAGEMENT

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Mr. Sanderford is President of MARSAND, INC., a hands-on technical consulting company. Services include turn-key installations, FCC Proofs, RF troubleshooting, transmitter and studio upgrades, personnel training and systems support. His exposure to broadcast facilities through extensive traveling over the past 25 years has prepared him to deal with the various aspects of the station engineer. The need for constant training and understanding of the financial and management requirements is underscored in this article. The simple, obvious, but often forgotten basics are brought to light in this presentation.

For the past several years now, the term "Bottom Line Management" has become familiar to everyone from top level management to entry level hourly employees. Management has used it to focus in on operating costs and trimming station operations. As engineers, we are responsible for the technical portion of the operation and must play the role of managers in it. The purpose of this paper is to highlight pertinent areas of management for the engineer as it applies to the station operation and provides some guidelines and examples of ways to manage more "cost effectively."

There are three main areas that contribute to managing the bottom line. These are **People**, **Time**, and **Money**. Although libraries hold volumes of books on each of these areas, we will be discussing only the practical and relevant aspects of each. The principles and applications of this paper apply to all levels of engineering -- from Technician, to Supervisor, Chief Engineer, Director of Engineering, and the Contract or Consulting Engineer.

PEOPLE

Contrary to the typical engineers' perception, non-technical people are not moronic or incapable; instead, they are untrained, unchallenged, and/or unmanaged in specific technical areas. Today's engineer must learn to adapt to and communicate with fellow workers to a far greater extent than in the past. Oddly enough, we are in the business of communicating to the public, but we do little to understand and communicate with each other. Why don't people understand what we say or do? Consider the following list:

* Communication is established when the **OTHER** person **UNDERSTANDS** what you are saying. Merely making a statement, writing memos, or putting it in writing in front of the equipment does not guarantee that the idea has been communicated. Consider the person you are addressing and make the effort to communicate within that individual's realm of understanding. You may be absolutely right in what you say, but it will have no meaning if the listener does not know what you are talking about.

* Check your communication **effectiveness** by sampling a response. Involve the person during the process by asking questions, comments, input of ideas, or even their version of what you just said.

* **Communicate** in a clear, objective, and professional manner. Assumptions and insinuations are the lowest forms of communications. To quote an old adage: "Say what you mean and mean what you say." Be prepared and know what you want to say, and evaluate the ability of the person to receive under the circumstances at that time. As you become objective and able to communicate your point, you will find that your ideas and proposals will be taken more seriously.

* **Be yourself**, or rather, the better side of yourself. Putting on airs, talking down to someone, speaking in a demeaning way or in an arrogant manner only generates a level of intolerance and resentfulness -- neither of which is conducive to good communication.

* **They are Human, too!** Never assume that the other person interprets what you do in the same exact manner. What you understand and experience is your own; do not expect someone else to receive it with the same enthusiasm and level of understanding. You can, however, by considering the other person, convey the importance, excitement, seriousness, or novelty to an equivalent level through their own understanding. If someone asks you for a piece of paper, do not waste their time explaining how it was manufactured.

* **Learn to listen!** Speeches and sermons are one way conversations. Communication is the art of knowing when to talk and when to stop talking (an action commonly known as listening). It is amazing how much one is able to learn when the listening mode is turned on. In the world of data communication, even the bytes know the meaning of hand-shaking, which is their version of waiting until the other has stopped talking. "Even a fool, when he holds his peace, is counted wise, : ..." Proverbs 17:28.

* **Personal appearance** conveys the image of how you see yourself to others. Breath mints and deodorant can be called "man's best friend" in close encounters with people. Current dress styles take a back seat in comparison to neatness of personal appearance.

By no means do the above statements encompass all communication techniques, but they do establish guidelines that, if followed, will go a long way to successful communication. These areas were highlighted because they represent the most common complaints against the engineers. There are a number of good, short, entertaining, yet extremely effective books and tapes that deal with Communicating. One such tape series that I recommend is titled "Secrets of Power Negotiating" by Roger Dawson from the Nightingale-Conant Corporation, Chicago, IL. The title hides the fact that it contains some of the most valuable communicating skills I have encountered and deals with them in a very humorous, practical way.

Good management is the ability to consistently bring out the best product performance with minimum failures in the least amount of time at the lowest possible cost. Proper management of people is fitting each person into the place where they can best use their abilities to improve the company and meet the company's goals. Management of people is a constant state of negotiations. The most effective and long lasting form of negotiations is where both sides win. Here is another list of tips for managing and negotiating:

* A pat on the back, an **encouraging word**, a compliment or recognition, (great or small), is far more effective than disciplinary reviews or bear tactics for improving the performance of personnel.

* Make sure the person **understands** what **performance** you expect. Setting goals and progress checkpoints are very effective and allow for self-measurement. Training must be at the same level of job competence if proper performance is expected. Don't expect performance if the skills or use of tools and equipment have not been prepared through proper training.

* Develop and show trust and confidence in the people you work with. As confidence builds, so does good performance.

* **Be a team player**. Learn from the expertise of your manager and successful sales personnel. They are working to bring in the money that your department is spending. They are doing what they do best, just as you are. Keep your company loyalty at all times and in all places. Talking against your fellow workers, supervisors or employer will raise questions about your own credibility since you have chosen to work there.

* When working with problems, **always deal with the facts**. Just as basic troubleshooting techniques work with equipment, collecting solid data, variations from the norm, etc., should be used, especially where personnel are involved. Emotions and personalities are intangible and change according to circumstances. The specific problems they generate in affecting the workplace as they relate to performance, errors, insubordination, or violations can be itemized and evaluated as FACTS. Deal with the facts to yield the proper solution. Whether negotiating or following-up on disciplinary action, facts become indisputable items where emotions and perceptions are subject to individual interpretation.

* If you encounter a person without a smile -- give him one of your own.

* When negotiating or avoiding an argument, keep in mind that it may be possible for both sides to win. This little known tactic could keep a project on track or salvage an otherwise irreversible situation.

Communication and the management of people directly influence the performance of the station, your department, and ultimately the bottom line.

TIME

Whether we like it or not, we are all managers of our time. Some of us are good time managers, while the rest of us only take time to wonder where the time we had went. One of the most noted reasons for high levels of frustration is poor management of time. Poor planning (or none at all), deviating from goals, interruptions, emergencies, unnecessary work and day dreaming all add up to wasted, unrecoverable time. I am not advocating a robotic process to the extent of accountability minute by minute. There are, however, many advantages to examining one's activities and evaluating them in light of efficiency and necessity. Some of these advantages are lowering the frustration level of work and actually accomplishing goals. No matter how much time and effort of your own you spend, the property you are responsible for belongs to someone else -- take care of it in a responsible, professional manner. Here is a list of items to look for:

* Take time to organize your day. Use a simple legal pad, title it with the day/date, then list the activities and goals, as individual line items, for that day. On items

requiring additional information from later responses, skip lines to allow room. Keep each day on a separate sheet and generate a file folder for each year. A computer with a scheduler can also be used in the same manner. Be sure to make a hard copy after editing at the end of each day. This planning can be done at the end of each day in preparation for the following day or at the beginning of the day, **before** work activity begins. Check off the items as they are completed. Review the previous day's notes for unfinished items and add them to the daily list. If items are not completed, transfer them to the following day and so on until they are.

* How do you use your time? Once you make up your mind that you **really do** want to know where your time goes, be ready to invest a disciplined, revealing effort for one entire week. Have a watch and notepad with you the entire time. Every quarter hour, make an entry as to what you did during the past 15 minutes. Include phone calls, interruptions, breaks, trips to various locations, reading time, meditation, conferences, listening to gossip, generating gossip, answering questions, troubleshooting equipment, etc. At the end of each day, look at all the information on the notepad and find out where your time has gone. Once a picture of your work day is captured, make a decision to do something about it; after all, you spent all that time and effort to find out where it went. It is your time, isn't it?

* Find out when your most productive time is. Only <u>you</u> know when you operate at peak efficiency. Block that time of day for your most intensive work.

* "**NO**" is a very important word in the English language that is improperly or seldom used. Do not be afraid to use it. Use it to manage your own time. Learn to say no to people or activities that would hinder your work performance. Using "NO" firmly and wisely will eventually carry authority and respect with it. Do not use it arrogantly or thoughtlessly. Remember the communication skills, listen before you respond. Be flexible and adaptable, but always steer back on course to your daily outline.

* FILE, FILE, FILE! Learn how to document and file. One of the most dreaded words in the engineering world is <u>paperwork</u>. Without it our ability to retrieve information or produce credible data would be impossible. Paperwork is not so bad if it is kept to a minimum. Learn to write brief sentences. Use the daily planner to keep track of all calls, in and out, along with appointments and incidents. Learn to file effectively by getting rid of outdated and duplicated material. Keep only active and reference files nearby. File outdated, but relevant information in storage. These files may be useful should any legal actions take place. Develop a system that others can logically follow. Someday you may call to have someone try to find a crucial item for you.

* Apply the time efficiency test to your department. Arrange schedules and activities to minimize unnecessary overlaps and wasted time. Work with technicians and operators to understand the sequence and motions of activities they perform. Communicate with them and receive input in determining better ways to perform the job functions.

Time management is a crucial ingredient that affects the bottom line. A good book to read is "The One Minute Manager." The principles set forth are very useful and help explain, to a great extent, the more modern methods of management of highly successful corporations. Much of the discussion in the book is directly applicable to the engineer in the management of time and people. Keep in mind that at some point, delegation ceases and someone has to perform the basic task.

MONEY

Last, but certainly not least, is the management of money as it relates to the engineer and his/her department. Cost effective engineering is the ability to properly manage the purchase, repair, and maintenance of equipment, operations, personnel and utility costs. They are called costs, implying negative income, due to capital equipment and maintenance expenditures. The engineering department seldom has opportunity to generate revenues other than tower space leases or remote equipment use. Traditionally, the engineering department has been (and still is) looked upon as major financial drain of the station. (Although some News departments and Programming costs may run close competition.) For years, engineers have tried, and eventually succeeded in showing managers and station owners their efforts in implementing maintenance budgets, optimizing the use of personnel, updating equipment for better performance and less power consumption. This was being done even before the "bottom line management" became a buzzword. Unfortunately, the efforts of these engineers were not shared with their peers to the degree of becoming a norm in their stations. Articles are now being written in the trade magazines that are causing engineers to wake up and pay attention to the changing role of the engineer in today's society. These articles are pointing out problems with solutions, trends, and guidelines that alert engineers who want to stay in the business of broadcasting would do well to heed. The following is a guide for checking out your department:

PURCHASING EQUIPMENT:

* Is the item a need or a want?

* Is the item to correct an existing problem, replacement, or enhancement?

* Is the item maintenance compatible with existing equipment? There are times when it is expedient to keep the same model or manufacturer type in order to make use of spares and alignment tools. A newer, although better, unit by another manufacturer might generate additional maintenance and parts problems.

* Will the same format be maintained? Adding a more efficient system may look a little better over the air, but may generate excessive hidden costs if a different format is used from the existing one in the station. Dubbing and transfer losses, mixed tape purchases, sizes of storage shelves, etc., can generate confusion and limit the interchangeability of tape and equipment.
* Is the item in the budget?

* Does it follow goals and objectives of the station?

* Who asked for the item? Production? Sales? GM?

* Is there alternate equipment to be used should purchase be delayed?

* Is there training required in the maintenance and/or operation, and at what cost?

* Is the level of technical expertise available on staff or by contract to maintain the item?

* Are the shipping and taxes included?

* Is this the best value? Have all sources been evaluated equally in light of the function they will perform? The lowest price can turn out to be more expensive than one of best value. Too often, purchasing or accounting look at price with no regard for value. Equipment value is an engineering decision and is based on function, productivity, maintenance time and cost, equipment life span,

conditions of use, obsolescence and technical support. Getting what you pay for is a fair description of this highly competitive and rapidly changing technology.

* Does it meet environmental EPA and OSHA standards?

* Is there a place to put it? Think of rack space, special requirements such as clean air, extra cooling, additional power source, table top area, glare from lights.

* Personnel safety?

* Surge protection or backup power source UPS?

* Special permits if building modifications are required?

* What about Warranty? Read the fine line!

* Does the item have a track record? Many times newer is not necessarily better if it made it on the market before all the bugs were worked out. What is your risk factor?

REPAIR

* How do the repair parts compare to a new unit? Always include the time it takes to perform the repair.

* How many times has it been repaired? A maintenance folder on each equipment item showing problem, repair, parts used, date, technician, etc., is an excellent record to evaluate an item at any given time. This type of documentation is very effective in purchase request environments.

* Can the equipment be modified or upgraded to avoid the same problem in the future? Who else has already done it?

* Is the item still serving a useful function? There is no need to repair an item that is no longer going to be used or is near extinction. Look for ways of getting rid of unused or obsolete equipment by means of donations, selling, or plain trash. The trend for Municipal taxing authorities is to tax all inventoried items whether they are usable or not as long as they are on the property. High cost items that have been depreciated out on the books for IRS purposes may still be taxed at rates as high as 30% of the original purchase value, each year with no depreciation, for as long as it remains in the station.

* Can it be patched together to live yet a little longer?

* Can the repair be made by local contract engineers or technicians at other stations? Sometimes pooling or networking in the technical community can be a great asset to all involved. Back-up, expensive test equipment, factory trained expertise, and the opportunity to discuss problems and share solutions become available as invaluable resources.

PROJECT TRACKING

* Is there a time table? Most project oriented schedulers help prioritize and project visual aids, such as the Ghant chart.

* What are the budget and item cost breakdowns?

* Is there a commitment on delivery? Do the suppliers understand the need for timely delivery? Have they been instructed to notify you of any changes?

* Have allowances been made for shipping time? Method of delivery?

* Is there a list of alternate sources for each item should commitments not be fulfilled?

* Insurance for purchases? Contractors?

* Are payments on schedule?

* Who is in charge? Next in line?

* Who can make changes? Who should be notified? Procedure?

* Documentation?

* Permits?

A project cost accounting system using a spreadsheet with formulas tied to cells to keep track of total sums is a simple and inexpensive resource. A financial accounting is instantly available at any given time during the project.

A program currently available for Maintenance Management is BCAM by Computer Assisted Technologies from New York. BCAM is a database oriented software that sets up maintenance schedules, procedures, document repairs, parts and inventory, provides troubleshooting techniques and upgrade information on equipment, shift schedules, contacts -- in short, manages the majority of aspects that the engineer deals with on a daily basis. It is very user friendly, easy to input and extract information. It has further flexibility to be integrated with VidCad, the cable video and audio documentation software that has become the leader in the industry. These two programs along with the low cost of computers can provide the engineering department with the necessary tools to meet the bottom line requirements as demanded for today's cost effective engineer.

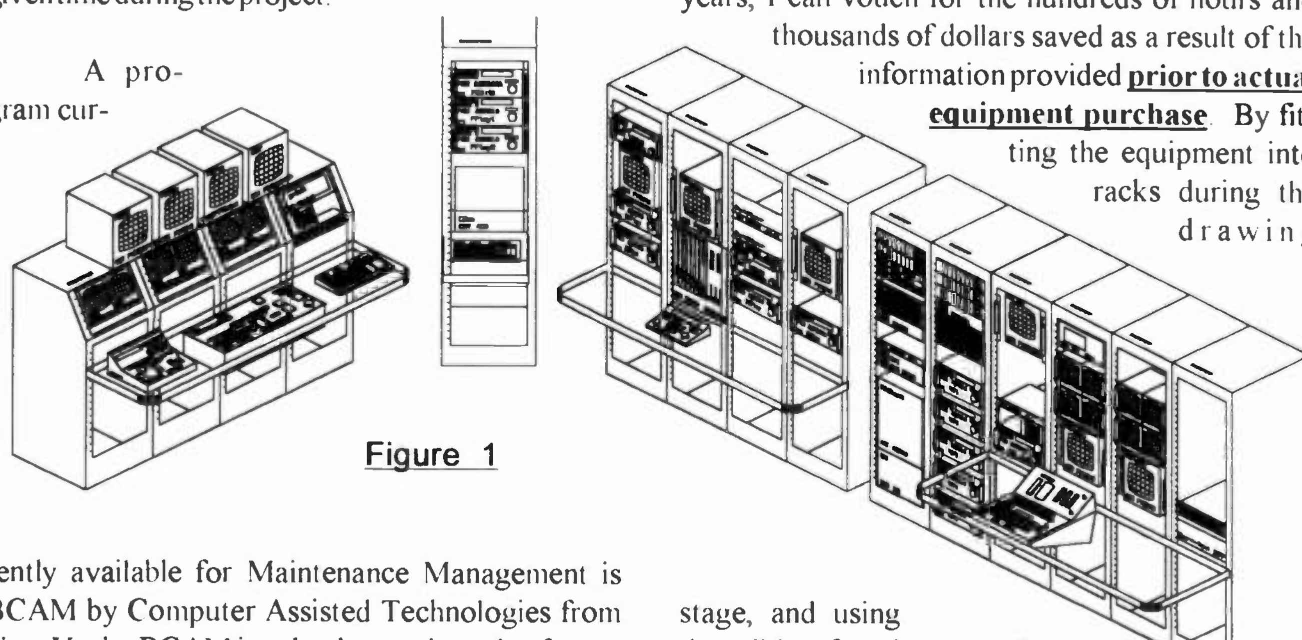

Figure 1

Figure 1 is an equipment rack layout drawn to scale and in 3-D. Figure 2 is a portion of a wiring diagram for the same rack layout. Figure 3 is the cable run list for the wiring diagram. All of these were drawn using VidCad and it's equipment libraries and routines. Additional information automatically available include equipment lists, individual and cumulative lengths of cable, connector quantities and types, cable labels preformatted or available for customizing, and signal path tracing -- to name only a few. Having used this program for over 5 years, I can vouch for the hundreds of hours and thousands of dollars saved as a result of the information provided **prior to actual equipment purchase**. By fitting the equipment into racks during the drawing stage, and using the editing function to shift items around, I was able to ergonomically place each item for optimal effectiveness. This was certainly a lot better than waiting for it to arrive, place it, find out it needed to be elsewhere, shift equipment physically around, etc. -- most of us have been there. Missing items in the preliminary equipment list are immediately spotted in the wiring program as point to point hook-ups are made. Cost effective engineering? Use the good resources being made available to you.

All of the preceding discussion has a common thread in managing the bottom line. People, time and money are all interrelated. Paperwork, the albatross of the engineer, has become essential. Generating a "paper trail", a term graciously donated from the accounting gurus, is now a normal requirement. Accountability has come to the forefront. The engineer's technical jargon intended to keep management at bay has lost its effectiveness and is being replaced by specific requests with measurable costs and achievable goals.

The ideas presented in this paper have been simplistic in nature. All of the listed suggestions have come from practical application and found to work very successfully when applied in a consistent and organized manner.

As in every form of business, each station is managed in a unique way. Most stations have budgets, some do not. Sometimes engineers are involved and have input in the preparation the budget. Some engineers are very fortunate to actually have control over the making and implementation of their budget, while others have budgets handed to them that are unrealistically low. Cost effective engineering is a team effort with good communications between the General Manager and the engineer and proper management of resources -- people, time and money.

Figure 2

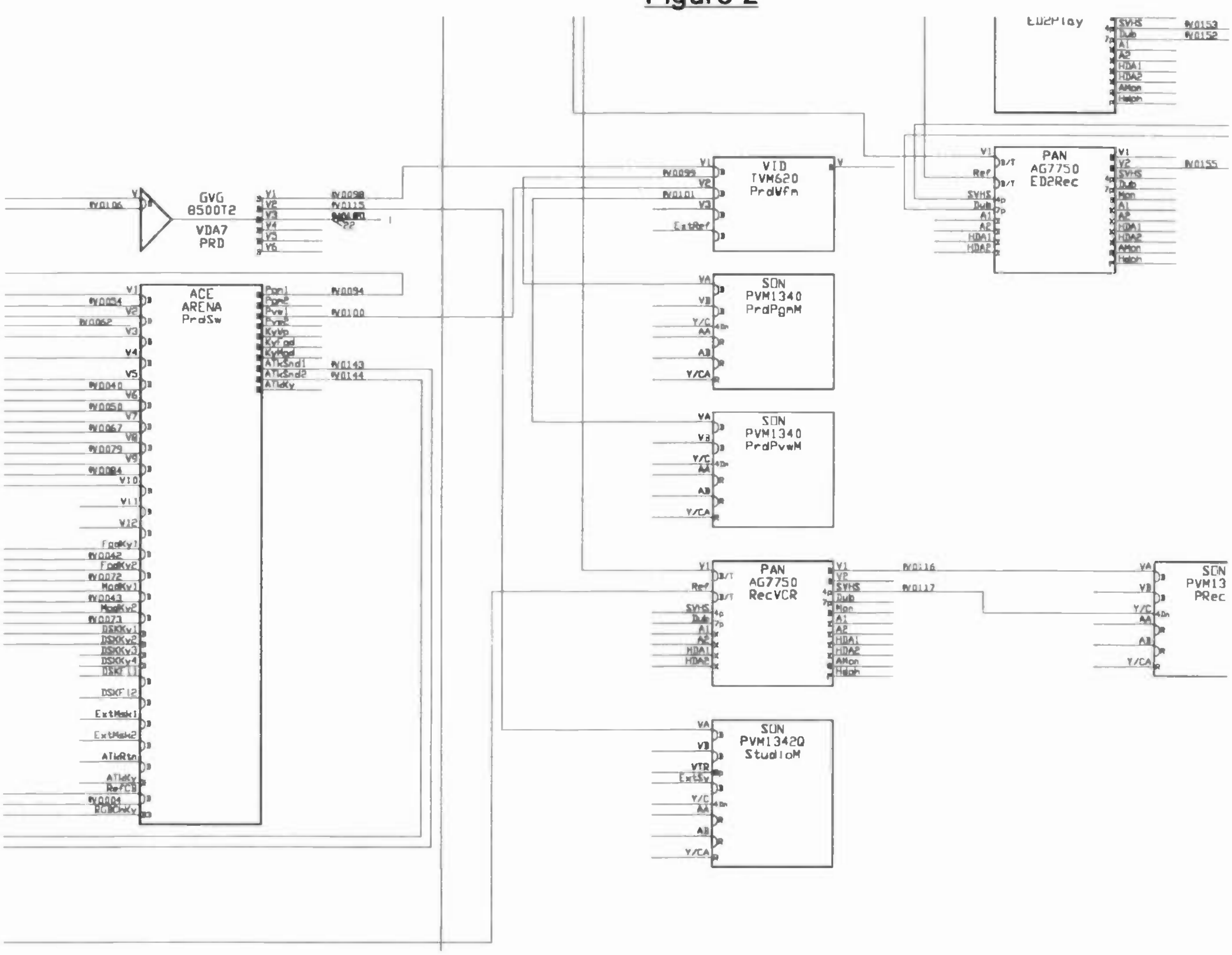

Figure 3

Cable Run List (Format 0)

Cablenumb	Source	Sourceout	Dest	Destin	Srk	Drk
0000	Source	SourceOut	Destination	DestIn	SourceRack	DestRack
A0001	ADA-1L	A1	AJPL2	A01	R9	R7
A0002	ADA-1R	A1	AJPR2	A01	R10	R7
A0003	ADA-2L	A1	AJPL2	A02	R9	R7
A0004	ADA-2R	A1	AJPR2	A02	R10	R7
A0005	ADA-3L	A1	AJPL2	A03	R9	R7
A0006	ADA-3R	A1	AJPR2	A03	R10	R7
A0007	ADA-4L	A1	AJPL2	A04	R9	R7
A0008	ADA-4R	A1	AJPR2	A04	R10	R7
A0009	ADA-5L	A1	AJPL2	A05	R9	R7
A0010	ADA-5R	A1	AJPR2	A05	R10	R7
A0011	ADA-6L	A1	AJPL2	A06	R9	R7
A0012	ADA-6R	A1	AJPR2	A06	R10	R7
A0013	ADA-7L	A1	AJPL2	A07	R9	R7
A0014	ADA-7R	A1	AJPR2	A07	R10	R7

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ACKNOWLEDGMENTS

A special thanks go to my son, David, a Senior Electrical Engineering major at Georgia Tech. University, and Dr. Walter Black of Video Design Pro for their technical critique of this article.

DAB I - TECHNICAL CONSIDERATIONS FOR DAB PERFORMANCE

Monday, April 19, 1993

Moderator:
Don Wilkinson, Fisher Broadcasting, Seattle, Washington

MULTIPATH PROPAGATION AND FADING STATISTICS FOR DIGITAL AUDIO BROADCASTING IN THE VHF AND UHF BANDS
Kenneth D. Springer
National Association of Broadcasters
Washington, District of Columbia

***EIA MULTIPATH TESTS IN CHARLOTTE, NORTH CAROLINA**
Bob Culver
Lohnes & Culver
Laurel, Maryland

***DAB: A RECEIVER MANUFACTURER'S VIEWPOINT**
Almon Clegg
Consultant
Rockaway, New Jersey

***TERRESTRIAL DIGITAL RADIO BROADCASTING AT L-BAND: PRELIMINARY RESULTS OF FIELD TESTS**
Francois Conway
CBC
Ottawa, Ontario, Canada

PROJECTED CONVERSION COSTS FOR DIGITAL AUDIO BROADCASTING
Skip Pizzi
Broadcast Engineering Magazine
Overland Park, Kansas

*Paper not available at the time of publication.

MULTIPATH PROPAGATION AND FADING STATISTICS FOR DIGITAL AUDIO BROADCASTING IN THE VHF AND UHF BANDS

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ABSTRACT

A number of parameters for quantifying a multipath propagation environment have been developed. Among the most important parameters for a digital audio broadcasting system are: amplitude fade depth, amplitude fade duration, spatial correlation of amplitude fades, channel delay spread, and channel coherency bandwidth. Multipath characteristics are heavily dependent on the channel environment and these parameters are therefore best quantified statistically. The literature reveals that these parameters have been measured in a number of settings and statistics have been developed. These statistics reveal that amplitude fades of greater than 18 dB are not common. Deep amplitude fades are relatively long (at least several seconds) for a stationary receiver. Delay spreads are rarely greater than 6 μsec in urban settings. And channel coherency, as defined in this paper, is rarely greater than 300 kHz. In designing a multipath mitigation strategy, it is recommended that the statistics revealed in this paper are applied.

INTRODUCTION

Many organizations are developing systems for digital audio broadcasting (DAB) which will operate in the VHF and UHF radio bands. Key to the success of a DAB system is the ability to operate in a multipath propagation environment. This paper will survey and summarize the literature relating to multipath propagation statistics.

DISCUSSION

Multipath propagation presents a number of challenges to reception of digital audio signals. Delayed echoes caused by dispersion or reflection in the transmission channel can lead to high levels of intersymbol interference (ISI) and an irreducible bit error rate for high data rate transmissions. Destructive interference can cause amplitude fades which, in marginal signal areas, cause the receiver to make bit errors, perhaps resulting in muted audio output.

Techniques have been developed to mitigate these effects. Orthogonal frequency division multiplexing (OFDM) allows the transmission of high data rates through the channel while allowing each bit interval to exceed the delay spread of the channel. This reduces or eliminates ISI errors. Time, space, and frequency diversity, combined with error correction coding, can be successfully used to combat amplitude fading. Adaptive channel equalization is sometimes used to combat delay spread and amplitude fading.

For any digital audio broadcasting systems, there are a number of statistics relating to multipath propagation which are important in designing a multipath mitigation strategy. Chief among these are: amplitude fade depth, spatial correlation of amplitude fades, amplitude fade duration, multipath delay spread and frequency coherency of the channel. The literature reveals research into these statistics for urban, suburban, and rural channels. This paper will summarize that research. Additionally, the literature often distinguishes between multipath effects on fixed and moving vehicles. This paper will try to address multipath from a locational (fixed) point of view. If a vehicle is mobile, the speed of the vehicle can be factored into the reception statistics and accounted for in a mitigation strategy.

Fade Depth

Egli[1] indicates that, in general, large sector field strength variations in VHF transmissions follow log normal statistics. In contrast, experimental evidence (Nylund[2], Finger & Toreck[3], and Okumura[4]) reveals that the small sector field strength follows Rayleigh statistics, at least in urban and suburban areas. Egli[1] suggests that

Rayleigh statistics may not apply in wide open areas and that log-normal may be more appropriate in those circumstances.

The Rayleigh distribution indicates that the field strength at 10 % of locations will be at least 8 dB less than the local average field strength. At 1 % of locations, the field strength will be at least 18 dB less than the local average. Nylund[2] reports measurements in New York City at 150 MHz indicating that, at over various localized small sector areas, the field strength at 1 % of the locations was at least 16 - 22 dB below the local average.

Young[5] reports tests at 450 MHz and concludes that the average location experiences 0.01 echoes of 0 to -6 dB in magnitude delayed at least 7.5 μsec. from the main signal. A location will average 0.1 echoes of 0 to -12 dB magnitude with at least 4.5 μsec. of delay.

Spatial Correlation of Amplitude Fades

The spatial correlation of amplitude fades addresses the likelihood that a localized fade will extend over a given geographic area. Nylund[2] reports experiments at 150 MHz indicating that the average width of fades of 10 dB or more is 0.55 feet. 10 % of fades greater than 10 dB are at least 1.35 feet wide. Jakes[6] shows that the theoretical normalized envelope cross-covariance coefficient, ρ_r, is governed by the Bessel function:

$$\rho_r = J^2_o\,(2\pi x/\lambda) \quad (1)$$

which is zero at values of x/λ of 0.4, 0.88, and 1.38. For example, at 100 MHz, the spatial correlation of a localized fade would go to zero over a space of about 2.6 meters.

Amplitude Fade Duration

When considering fade duration, a distinction must be drawn between fixed and mobile receivers. Mobile receivers will experience time-varying fading simply because they are moving through areas with different signal levels. Fixed receivers will experience some time-varying fading, for example, when the predominant fading mechanism is atmospheric refraction as opposed to multipath reflections from fixed objects.

Atmospheric changes can be expected to be very slow in general, leading to amplitude fade durations of several seconds or longer. Faster variations can be expected when receiving a signal near the presence of moving reflectors such as automobiles. Nielson[7] reports that tests near a four lane highway in Palo Alto, California revealed a Doppler spread of less than 6 Hz. Thus, amplitude fades can be expected to last for very long periods of time, especially in more benign environments.

Multipath Delay Spread

Delay spread is a measure of the statistics on the time distribution of the arrival of echoes of various amplitudes at a given location. Cox[8] defined the **delay spread** as the standard deviation of the power delay profile. Power delay profiles reflect the geography of a particular location. Delay statistics tend to be characterized in terms of urban, suburban, and rural environments.

Young[5] reports that tests at 450 MHz in New York City revealed a probability of greater than 40 % that a location will experience at least one echo of 0 to -6 dB in magnitude from 0 to 3 μsec. after the main signal. Meanwhile, the probability is less than 10 % that there will be any echoes of 0 to -6 dB in magnitude longer with a delay of more than 5 μsec. For 0 to -12 dB echoes, the probability is greater than 80 % for at least one echo from 0 to 3 μsec. after the main signal, while it is less than 10 % for an echo with a delay of more than 8 μsec.

Cox[8] reports on multipath delay experiments at 910 MHz in New York City revealing delay spreads of 2 to 2.5 μsec. Cox[9] reports on subsequent 910 MHz New York City tests revealing that 90 % of locations experienced a delay spread of about 2.5 μsec. or less and 99 % of locations experienced a delay spread of about 3 μsec. or less. Turin[10] obtained similar numbers at 1280 MHz in San Francisco.

Nielson[7] reported multipath delay measurements at 1370 MHz in urban, suburban, and rural areas in the San Francisco bay area. These tests revealed average delay spreads of 1.63 μsec. and 2.5 μsec. for suburban and urban areas respectively. 99 % of urban locations experienced a delay spread of less than 3.6 μsec. The tests also showed that the probability in any environment of a multipath echo of amplitude between 0 and -20 dB rapidly approaches zero for time delays greater than about 5 μsec.

More recently, McLarnon, et. al.[11] report multipath measurements at 798 MHz around Ottawa, Toronto, and Montreal. In these areas, they report average delay spreads of 1.3 μsec., 0.8 μsec., and 0.5 μsec. respectively. 90 % of locations experienced delay spreads of less than 3.7 μsec., 4.2 μsec., and 2.1 μsec. respectively. The longest delay spread reported in 880 locations in these three cities was 8 μsec.

However, delay spread does not reveal the entire picture regarding the effect of multipath reflections in a given area. Cox & Leck[12] show that the effect of an evenly distributed set of echoes and a single, large amplitude echo will be completely different even though both situations produce the same delay spread.

The power delay profile is also a function of frequency. As frequency varies, the reflective characteristics change and hence the number and strength of multipath echoes varies. However, comparing test results noted above, spanning 450 MHz to 1370 MHz, reveals that there is little variation in actual delay spread results.

Finally, McLarnon[11] and other sources indicate that **delay interval**, defined as the length of time for which the channel impulse response is above some threshold, may be more important in characterizing intersymbol interference effects for a digital transmission.

Frequency Coherence (Fade Bandwidth)

There is little experimental data available on wideband multipath fading over localized areas. In particular, for digital audio broadcasting, the **coherence bandwidth** is important in assessing the statistics on the frequency bandwidths of amplitude fades. The 90 % coherence bandwidth is the bandwidth over which the correlation of instantaneous signal level is 90 %. This is a measure of the likelihood that a signal level fade will extend over the bandwidth.

Cox & Leck[12] report the results of a number of experiments at 910 MHz in New York City. The experiments revealed that fade coherence bandwidth at a given location is inversely proportional to the delay spread. However, the results also showed the importance of the actual power delay profile, i.e., two power delay profiles with the same delay spread may produce radically different frequency correlation patterns. In particular, a power delay profile which consists of a single, large reflection will produce a sinusoidal correlation pattern, while a delay profile with a smoothly decreasing amplitude vs. delay characteristic will produce a smooth frequency correlation pattern, even though both profiles yield the same delay spread. Obviously, delay spread can only be used as a rough surrogate for fading spectral correlation.

Cox & Leck[12] report that 10 % of locations showed a 90 % coherence bandwidth of greater than 300 kHz. Meanwhile, only 10 % of locations had a 90 % coherence bandwidth of less than 30 kHz. They plotted delay spread versus coherence bandwidth for a variety of locations. In finding a "best fit" curve to the distribution, they obtained a formula that:

$$B_{0.9} = 90/D \quad (2)$$

where: $B_{0.9}$ is the 90 % coherence bandwidth, in kHz,
and D is the delay spread in μsec.

Jakes[6] shows the cross correlation, ρ_f, of signal amplitude at two frequencies spaced by ΔF to be:

$$\rho_f = 1/[1 + (2\pi * \Delta F D)^2] \quad (3)$$

So, setting $\rho_f = 0.9$; ΔF, in kHz, becomes $B_{0.9}$ and, with D in μsec.:

$$0.9 = 1/[1 + (2\pi * B_{0.9} * D/1000)^2] \quad (4)$$

or,

$$B_{0.9} = 1000/(6\pi D) \approx 53/D \quad (5)$$

which shows fair agreement with Cox & Leck[12]. The difference between theoretical and experimental results could be attributed to the widely different delay profiles, including cases of a single, strong echo, reported by Cox & Leck[12].

Recent measurements at 97 MHz in San Francisco indicate that deep, broadband fades are rare. Measurements at 11 locations revealed a "worst case" fade bandwidth of 80 kHz for a 17 dB fade. Because, as was noted earlier, the power delay profile can change with frequency, so can the coherence bandwidth. More measurements of coherence bandwidth are needed, especially in the VHF band.

CONCLUSIONS FOR DAB SYSTEMS

One of the major improvements to be expected in a digital audio broadcasting system is the substantial elimination of audio degradation due to multipath propagation conditions. For a digital transmission, multipath can be thought to have two primary effects: (1) delay spread in the channel caused by strong echoes yielding intersymbol interference and, under some conditions, an irreducible bit error rate in the receiver; and (2) RF signal loss due to broadband amplitude fading.

There are well-developed strategies for combatting these effects. One popular approach to overcoming delay spread ISI problems is the use of OFDM, often with a small guard interval between bits to assure orthogonality. Adaptive equalization is sometimes used to combat both delay spread and amplitude fading in a channel, although its applicability in a mobile environment less certain. Meanwhile, amplitude fading is often attacked through combinations of spatial, frequency, and time diversity. If a receiver is fixed, multipath can often be mitigated by simply moving the antenna slightly. In a mobile receiver, the statistics of multipath propagation become important.

It will take many years to fully assess the success of a new DAB system in actual practice. Therefore, it is important to model a proposed system and evaluate how it is likely to perform against some "bounds" of real-world multipath conditions. There is a large amount of empirical data available but, unfortunately, much of it is not in the FM broadcast band. Fortunately, however, there is support for the proposition that most of the important statistical parameters are relatively constant across the VHF and UHF bands. From the literature surveyed here, it seems that the following numbers are reasonable bounds on multipath propagation effects:

99 % fade depth: 18 dB

Fade correlation area: < 2.6 meters

Fade duration: > several seconds

99 % delay spread: 6 μsec.

90 % fade bandwidth: 300 kHz

It is appropriate that DAB systems be designed with these numbers in mind.

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PROJECTED CONVERSION COSTS FOR DIGITAL AUDIO BROADCASTING

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As the transition to a digital broadcast format looms larger on radio broadcasters' horizon, the actual cost of such conversion takes on critical importance. For long-term financial planning and future marketing forecasts, the capital investment requirements that lie ahead must be identified. Although actual products are not yet designed, prototypes and projections for the several system models have been made, allowing reasonable approximations of conversion costs to be predicted.

This examination will consider the three general models proposed for *terrestrial* digital radio broadcasting, and also touch upon these scenarios' effects on the cost of regulation and consumer receivers. Incremental operational costs for broadcasters will also be discussed. Finally, differences in potential return on investment presented by these models will be explored.

Some initial caveats: not all of the digital broadcast models considered here have yet been proven workable under actual field testing conditions. Therefore, this analysis assumes, perhaps prematurely, that proponents' claims for their formats are indeed valid; it proceeds from that point to assess the relative financial impact of these proposals. Secondly, generalizations are made among actual proposals to group them into three global categories. For purposes of this examination, such groupings are considered appropriate.

Terrestrial models

Three significantly different approaches are now under scrutiny for terrestrial digital radio broadcasting. The first injects a digital signal into an existing, active broadcast channel in a compatible manner, such that both existing and new digital radios can receive their respective signals on the same frequency. This is the so-called *in-band/on-channel (IBOC)* model. The second approach uses currently unallocated or "taboo" channels in the FM band for digital radio broadcasts -- the *in-band interstitial (IBI)* methodology. Finally, an entirely new spectrum may be used for digital radio broadcasts in the *new band (NB)* approach (also called *out-of-band*).

Each of these methods will require different complements of equipment, regulation and operation to establish successful service. Variation in these parameters that the current group of format models provides is considered in the following analysis.

Transmission hardware

Reasonably safe assumptions can now be made about the equipment that each general digital radio broadcasting model will require at the broadcaster's transmission site. See Table 1 for a summary.

· *IBOC:* Contrary to earlier expectations, it now seems evident that the Class C amplifiers used in analog broadcast transmitters do not possess sufficient linearity to adequately handle digital signals in an IBOC model. This renders false the previously held tenet that an IBOC exciter *alone* will be required for transition to this system.

Therefore, in addition to the digital exciter, a separate low-power transmitter (a Class A or Class AB RF amplifier) will be needed for the digital signal. The output of this transmitter will be combined with the analog transmitter's output by an RF hybrid. The existing transmission line and antenna can be used for propagation of the combined digital/analog signal in most cases. Exceptions to this might involve FM stations that are currently transmitting through a combiner system, or possibly those using directional antennas. Even in such cases, modifications to existing hardware may be all that is required.

This approach is considered nearly certain to be applied in *FM*-IBOC formats. In the only *AM*-IBOC format proposed to date (USA Digital), development has not progressed to a level where full hardware specifics can be identified, but it can be speculated that a similar arrangement might be used.

Further examination by that proponent and others considers the antenna system bandwidth in AM *directional* systems. Because of the bandwidth limits in such arrays, not only might a separate transmitter be required, but a separate, collocated (and probably omnidirectional) antenna system may be needed, as well.

· *IBI:* From its very first proposal, the interstitial approach has mandated a digital exciter and low-power transmitter operating on the IBI channel's transmission frequency in the FM band. Most IBI formats require the use of an existing FM broadcaster's adjacent channel, and must originate from a collocated site. This implies that the existing transmission line and antenna could again be used in most cases, with the same possible exceptions as noted above for FM-IBOC systems. The digital exciter used in an IBI model may be less complex and therefore cheaper than an IBOC unit.

There are a few variants to the IBI model, however. First, the above case assumes an FM station will use one or both of its own first-adjacent channels for its digital signal. Some proposed systems allow for an FM station to use interstitial channels *other* than those adjacent to it. In such a case, it is likely that a separate antenna and transmission line will be required. Like all in-band systems, power requirements will be on the order of 100 to 1000 times lower than analog FM operating levels, so the antenna and transmission line will be less expensive than traditional FM systems. Rigid transmission line may not be required.

Further, under some scenarios, AM stations will be allocated FM-IBI channels for digital transmission. In these cases, the AM station will require a complete, new (albeit low-power) FM-band transmission system. There remains the possibility that multiple AM stations might share such a system, or that an IBOC/IBI hybrid arrangement might exist (whereby an FM station uses its own frequency in IBOC fashion, and leases its adjacent channel[s] to an AM station[s] for their IBI use). In either of the latter two cases, the cost to an AM station for IBI conversion would be reduced.

· *NB:* An NB system will require a complete transmission system, including (at minimum) an exciter, transmitter, transmission line and antenna. The power level of the transmitter will depend on the frequency band selected, but among the two possibilities under consideration worldwide (L-band and S-band frequencies), these powers might range from roughly equivalent to existing FM powers (L-band) to four or more times that amount (S-band).

TABLE 1

TRANSMISSION HARDWARE LIST & APPROXIMATE COSTS ($k)

ITEM	IBOC (FM)	IBI	NB
Exciter	$ 20	15	25
Xmtr	25	25	≥100
Combiner	5	5	
Xmsn line		(5)	30
Antenna		(10)	25
Monitor	5	5	10
STL/TSL & Rem. Ctrl.	(≥5)	(≥5)	50
Tower			(≥300)
Installation	5	≥5	25
Per-station factor	1*x*	1*x*	≤0.25*x*
TOTALS *(per station)*			
Lo	**$ 60**	**55**	**≤66**
Hi	**≥65**	**≥75**	**≥141**

( ) = possible expenses

The multi-kilowatt to multi-hundred-kilowatt transmitters required will likely be expensive to produce at these frequencies, as will the waveguide-style transmission lines and wideband antennas necessary. But the only NB format currently under consideration for terrestrial application (Eureka 147/DAB) employs a multiplex arrangement by which four or more program signals are combined in a single transmission system. Therefore, each station's cost of conversion to such a system is reduced by a factor of four or more.

Unlike either in-band model, an NB system will probably *not* be collocated with many existing facilities. If four or more stations are combined, a new transmitter site is likely to be required for at least some of the stations in every transmitter grouping. This will add to the cost of such a system, in both new site-acquisition expenses and new STL purchases. In some cases, the full cost of establishing a new transmission site from scratch (including tower construction) may have to be borne by the participating stations.

Consultant/legal costs may also be higher in an NB scenario because of the potential need for additional filings and propagation analyses required by new transmission sites and new spectrum usage.

The above assessment of an NB system assumes a single origination point, but this is not the only possibility here. A multi-transmitter, "cellular" approach may be used instead, employing many low-powered, co-channel transmitters placed within the market. This method takes advantage of the *single-frequency network (SFN)* element of NB format design. Among its advantages are better coverage and pattern control (allowing very specific tailoring the RF field to a market's boundaries) and a reduction in overall power consumption for a given coverage area. Its disadvantages involve higher startup costs for purchase and installation of the multiple transmitters, and higher continuing costs for the acquisition and maintenance of multiple sites.

While most of this discussion assumes an L-band (1.5GHz) application for terrestrial NB systems, the so-called *small SFN* system just described could minimize the relative power-efficiency disadvantage of the U.S.-preferred S-band (2.3GHz). If such a "cellular" terrestrial approach were selected, its application at S-band might be possible for an initial cost not significantly higher than at L-band.

· *All systems:* Under each transition scenario, some form of new RF monitoring equipment will be required by all stations. Additional audio chain improvements may also be desired by stations, in order to take full advantage of digital transmission quality. These could include new or upgraded STLs, improved studio equipment and new audio processing hardware. Additional services may also be made possible by a digital radio broadcast system, such as data transmission or a second audio service. New origination and STL hardware will be required to establish these services, in most cases. (Table 1 considers only RF hardware, and a simulcast programming scenario. Additional origination equipment may be purchased at a station's discretion, and such expenses would be relatively equivalent under any format model.)

Regulatory differences

The several transition models currently proposed will each present significantly different burdens to the FCC in establishing service. Although this cost is not borne directly by the broadcaster, it is an expense to the U.S. government, and therefore ultimately paid for by the American taxpayer.

Not all regulation-related costs are of the governmental variety, however. Digital radio service may allow the entry of new players, or allow competing applications to be filed. This may cause existing stations to incur additional legal fees and related expenses, and in the end may result in reduced revenue from increased competition.

Moreover, some observers of the regulatory process have also forecast that the additional cost of new regulation may be recouped from the licensees *directly*, in the form of spectrum fees or auctions for digital radio authorizations.

In relative terms, because the full AM/FM IBOC scenario will require no new authorizations, it will engender the lowest regulatory costs. After rulemaking establishes service, the authorization process for stations will require relatively minimal effort. An IBI model that stipulates adjacent-channel authorizations will be the next level of burden to regulators, presenting a significant increase in effort from the full IBOC approach. A more complex IBI system (incorporating all AM stations and allowing non-adjacent FM allocations, for example) could require even more regulatory work, as will any NB system. (See Table 2.)

It is well known that an IBOC approach also minimizes the likelihood of marketplace upheaval in the radio industry. An IBI or NB system in which all existing stations are summarily granted a "simulcast" channel (similar to HDTV plans), and no new allocations are involved should provide a similar outcome. But any non-IBOC service will likely bring with it a licensing process that places additional obligation on both the regulator and the applicant.

Receiver cost

The cost of digital radio receiving hardware will almost certainly start high and drop dramatically as manufactured quantity increases and economies of scale run their course. Actual prices are still largely speculative, but most proponents expect to reach a $200-$300 price point when full production levels are achieved. Introductory prices may be three to four times higher, however.

Some generalized projections of the transition models' relative receiver costs can be made, based upon the expected complexity of their design. Because of the wideband, real-time decryption and filtering required by IBOC systems, these may be the most expensive to implement. The nature of this approach mandates that the bulk of system complexity cannot be transferred to the transmission end of the system, as is generally desirable.

IBI receiver systems may be slightly to significantly simpler (and therefore cheaper) than IBOC designs, because they do not require a signal extraction process. Critical filtering will still be required in IBI receivers, however. NB receivers will be wider in bandwidth and require relatively high real-time processing power, but they, too, may not be excessively expensive to produce. For example, while IBI and NB receiver designers expect to use silicon-based circuitry, one IBOC proponent projects the need for a potentially more expensive gallium arsenide (GaAs) chip as the basis for its receiver. (See Table 2.)

Operating costs

Assuming simulcast programming is used, an existing broadcaster's primary incremental expense in operating an additional digital radio service is its impact on electrical power consumption. Added maintenance costs and technical expenses incurred in providing any new ancillary services are a secondary consideration here.

Both IBOC and IBI systems operate in the existing broadcast bands, and can therefore take full advantage of digital broadcasting's power efficiency. The increment to an existing broadcaster's power consumption should therefore be negligible.

Some formats (mostly IBI types) even allow for the eventual phase-out of the existing analog broadcast signal, if and when its listenership dwindled to levels unworthy of continued service.

On the other hand, single-origination-point NB systems will require high-powered transmitters, resulting in a probable electrical power consumption similar to that used by today's FM transmitters. Under current

TABLE 2

RELATIVE COST ASSESSMENT FOR DIGITAL RADIO SYSTEM MODELS

	IBOC	IBI	NB
B'caster cost	L	L/M	H
Consumer cost	H	L/M	L/M
Governmental cost	L	M/H	H
Operating cost	L	L	M
Transition time	L	M/H	H

(L = Low; M = Moderate; H= High)

proposals, however, this power bill is shared by four or more stations, reducing its impact to each operator. Multi-origination-point (small-SFN) NB systems might *further* reduce overall power consumption costs by as much as 3:1.

Consider also that certain non-technical costs may increase in the digital radio environment. Most notably, the U.S. Copyright office and the recording industry favor the introduction of *performance royalties* to be paid by broadcasters for the privilege of playing published music recordings on the air. This payment would come in addition to the existing *composer* royalties that broadcasters already pay through ASCAP and BMI. So far, the U.S. Congress has repeatedly refused to enact this royalty into law, but its proponents plan continued pursuit of such compensation. The transition to digital radio broadcasting presents an arguable opportunity for establishing the precedent.

Finally, if regulation and system design permit (and receiver penetration warrants), analog/digital simulcasting could eventually be replaced by initiation of separate audio services on the two channels. Cost of new program acquisition, studio space and staffing for the second service would then be incurred, of course, and this could add considerably to operational costs. Multitask-capable automation systems could significantly reduce the incremental studio and staffing costs for establishing a second service, however.

Potential for return on investment

Just as the cost of transition to each digital radio broadcasting model differs, the effective return to the broadcaster from that investment also covers a wide range. The potential benefits of each system should be scrutinized by broadcasters in this respect, as the formats become more clearly defined. Capacity, robustness and flexibility are all desirable attributes, and they can influence the long-term profitability of operation in a digital delivery system.

The competitive effects to broadcasters may also vary among models. The degree of parity achieved between existing AM and FM stations in the various proposals is a prime example. For NB systems, the possibility of new competition (including satellite-delivered signals) puts a very different complexion on the radio marketplace. The dislocation potential of NB systems may also have an effect (either positive or negative) on some stations, as they move to their new joint digital transmission sites.

Further fragmentation of the radio market has been a concern among broadcasters facing the digital transition. Some favor permanent simulcasting, with digital service acting purely as a qualitative enhancement. Others prefer quantitative growth, allowing second audio services to be provided after an initial simulcast period, once sufficient numbers of digital receivers exist in the marketplace. This could stimulate further growth in the popularity of digital receivers (because new, desirable services are unavailable without them), and synergistically reduce receiver cost further, as in the AM-to-FM transition of the 1970s.

Economies of scale in producing second-service signals are similar to those now under study by stations exploring duopoly arrangements. Unlike today's duopolies, however, such consolidation in digital radio second-service would be a *progressive* rather than *regressive* migration, in that the same number of players produce more services, rather than fewer players producing the same number of services. Similar economies are achieved, but more services to the listener result in the progressive approach. Actual aggregate listenership (ratings and TSL [time spent listening]) may actually *increase* -- both overall, and for individual players -- due to higher listener-satisfaction levels (e.g., a single operator's two new services may have a greater *combined* listenership than the station's previous single format). Therefore, an increased number of channels might result in an outcome contrary to that expected by many of today's broadcasters.

Conclusions

Exact transition costs for digital radio broadcasting cannot yet be specified, but relative trends among general format models have been established. As Table 2 illustrates, there is no overriding favorite from a cost perspective (although IBOC shows a slight edge at present), and cost is certainly only *one* among many issues to be considered in the choice of format.

Remember also that a *single* system from among these choices may not be adopted; rather, a merging of models or a model not yet identified may actually be selected. Consider the example mentioned above in which FM stations use an IBOC format while AM stations use IBI; here, technical parity (or near-parity) between AM and FM stations benefits the consumers, and a single radio broadcast band is established. Parity's impact on the broadcast marketplace might be countered in this model by allowing FM stations to retain their original FM allocation indefinitely, while AM stations must give theirs up after a phase-in period. This sort of technical and political horse-trading could have dramatic effect on the eventual outcome, and such matters are far from settled.

Meanwhile, Table 1 shows that the actual range between extremes for costs of new hardware among transition models may be narrowing, although just how a broadcaster should react to this data remains unclear. For example, under this assessment, an NB system shared by four stations might not cost an individual station much more to implement than an IBOC system (in a best-case NB scenario). Even with a higher price tag, though, the NB system might carry a greater potential for return from additional services and coverage it may allow. But increased competition in the NB scenario (from satellite services, perhaps) might erase this advantage. And so it goes. The factors to be weighed are obviously complex and still highly uncertain.

Speed of transition is also a factor (see Table 2). The faster the broadcaster and consumer make their conversions, the sooner the broadcaster will see a return on the investment and incremental operating expenses of digital radio broadcasting. This speed will be strongly influenced by the cost and perceived benefits of transition for both consumers and broadcasters (i.e., lower costs and higher perceived benefits will increase speed.)

As more becomes known, understanding of these issues should be continually reassessed, allowing the radio industry to make properly informed choices in the momentous deliberations that lie ahead.

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DAB II-IMPLEMENTATION ALTERNATIVES FOR DAB

Monday, April 19, 1993

Moderator:
Ken Springer, NAB, Washington, District of Columbia

***THE EIA DAB TESTING CRITERIA**
Tom Keller
Consultant to EIA
Springfield, Virginia

***THE NASA-VOA DAB SATELLITE RECEIVER DEVELOPMENT PROJECT**
Don Messer
VOA
Washington, District of Columbia

***PARAMETRIC STUDY OF THE COFDM SYSTEM**
Louis Thibault, Minh T. Le and Eduardo Casas
Communications Research Center
Ottawa, Canada

DIGITAL SOUND BROADCAST WITH AUXILIARY OVERHEAD CONTROL
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AN IN-BAND, ON-CHANNEL FM DIGITAL AUDIO BROADCAST SYSTEM
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*Paper not available at the time of publication.

DIGITAL SOUND BROADCAST WITH AUXILIARY OVERHEAD CONTROL

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Abstract:

The use of an Auxiliary Overhead Channel (AOC) to extend the applications and improve the performance of Digital Sound Broadcast (DSB) is examined. The issues of migration to new and/or enhanced services, of on-channel performance capability, and of robustness to transmission impairments through the use of the AOC are discussed. An AOC protocol is outlined and some variable data-rate features are suggested. A specific multitone implementation of DSB transmission with an AOC is also presented.

1. Introduction

Digital Sound Broadcast (DSB) offers the potential of several new broadcast services in the existing FM radio frequency spectrum, including CD-quality compressed audio, digital information services, message services, and possibly heavily compressed medium-quality video signals. The markets for these and other services are not yet defined, so it is desirable for any deployment of DSB to have an embedded ability to conform to those applications that the market determines as desirable. One method for implementing this ability is through the use of an Auxiliary Overhead Channel (AOC).

1.1 DSB services

The first application for DSB in the U.S.A. will be the overlay of CD-quality digital audio onto analog FM radio programs. Various compression methods for a digital stereophonic audio signal are under evaluation, and a data rate of between 128 and 160 kbps has been shown to have excellent quality [1]. With the inclusion of forward error correction overhead to protect such compressed audio against the inevitable transmission errors, the required data rate for a single digital audio signal is assumed to be between 192 and 240 kbps. It appears that communication at these data rates at an acceptable error rate, and without interference either from or into existing analog FM will be possible.

Considerably higher data rates may be possible if a station elects to stop analog transmission and to use all its spectral allocation for digital broadcast. At these higher bit rates, considerable spare capacity will be available for the deployment of new services. One use of the AOC would then be to inform the radio receiver through a predetermined and simple protocol how much of the remaining data capacity should be extracted and forwarded to the associated output device that completes the implementation of the new service.

A number of potential applications can be listed:

1. **The broadcast of a 2nd digital audio signal** (could be of benefit in areas where spectrum is crowded).

2. **The broadcast of auxiliary digital data signals** (program listings and times)

3. **The broadcast of various public information services**, i.e., news, traffic, advertisements, etc.

4. **The transmission of private (leased and encrypted) information services,** i.e., paging, message forwarding, etc.

5. **The broadcast of video services**, including medium-quality video to accompany digital Sound, video browsing, etc.

6. **Cooperating broadcasts,** permitting higher data-rate applications or yet more robust transmission performance when two or more broadcasters can agree to coordinate, possibly with multiple ownership.

7. **Initially unanticipated services,** as long as they can be implemented with the assistance of the AOC.

The AOC would then need to mark certain segments of the received data stream for the intended application and the receiver would have a simple demultiplexing capability to separate the signals. A method for implementing both the AOC and more generally the transmitter and receiver with an AOC is discussed in Sections 2 and 3, respectively. The point of remarks on applications is that DSB can have a mechanism--the AOC--by which to provide for its future. The AOC facility could permit DSB to regenerate itself gracefully in new applications and services.

1.2 On-Channel Performance Improvements

The terms "on-channel" and, more recently, "overlaid" have been used to describe the simultaneous broadcast of analog and digital audio signals on the same FM carrier frequency while conforming to an appropriate power spectral density mask for that channel. In order to permit separation of the analog and digital signals it is inevitable that the bandwidth of the digital signal will extend beyond the central portion of the spectrum of the FM signal. Use of the sidelobes in this way could cause interference with and/or from another digital or analog signal on an adjacent carrier.

Three possible uses of the available bandwidth are shown in Figure 1. Potentially interfering stations are usually separated by 400 kHz, and the most common situation is shown in Figure 1a; the DSB signal can use the sidelobes from f_c-200 to f_c-100 and from f_c+100 to f_c+200 kHz. The case with no analog audio signal is shown in Figure 1b; this situation permits the largest digital data rates to be transmitted. Figure 1c illustrates the problem when there is a potentially interfering station removed by only 200 kHz; this adjacent analog FM signal forces the digital signal to occupy only the sidelobe on the other side of the carrier: that is, half the bandwidth it normally occupies. This type of half-channel transmission increases the desirability of low compressed audio rates such as 128 kbps [1].

In the case where there are potentially -interfering stations on both sides of the intended analog/digital station, another use of the AOC would be to tell the receiver's tuner to move to a different carrier frequency where digital bits would be available (either in the "single channel per carrier" mode of Fig. 1b or perhaps even multiplexed with another station's digital data).

The flexibility to use whatever bandwidth is available is achieved by using a modulation method called **discrete multitone***. As described in more detail in Section 3, the transmitter sends signals on any of up to 128 4-kHz wide subchannels, which span a maximum of 512 kHz. The subchannels can

*Also called, particularly in Europe, Coded Orthogonal Frequency Division Multiplexing (COFDM).

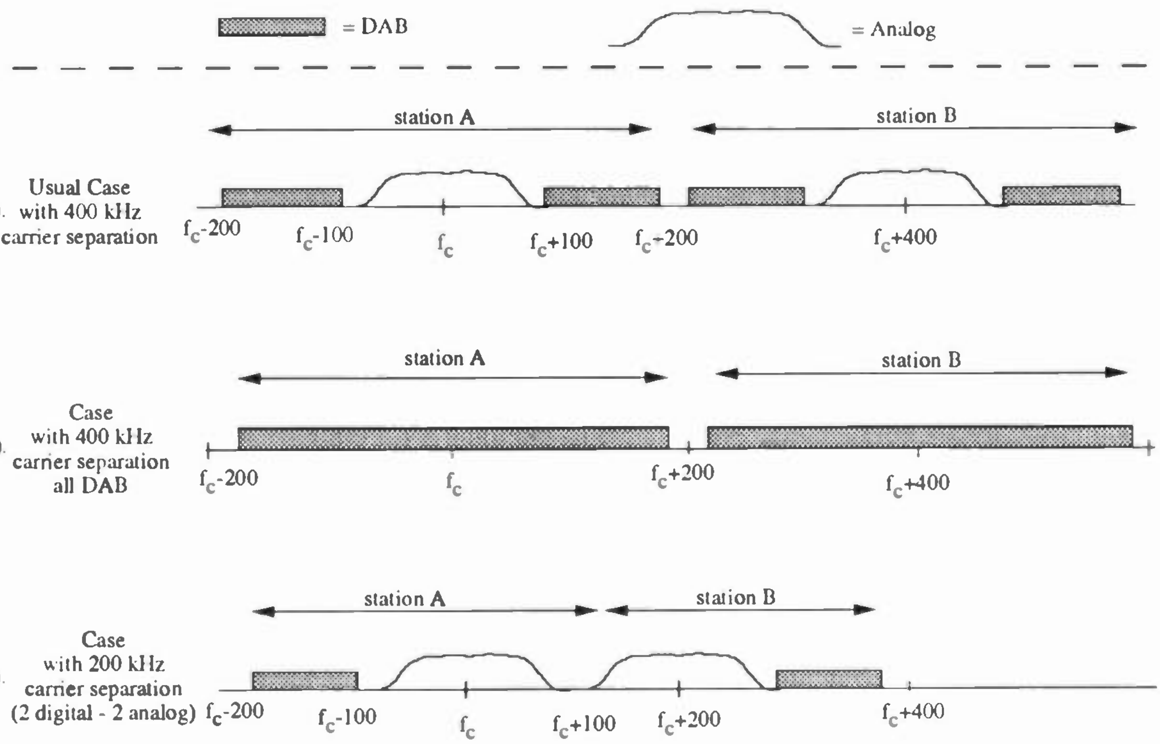

Figure 1 - Examples of Bandwidth Use with AOC

each carry between 1 and 5 bits per symbol (there are 4000 symbols per second), leading to data rates of 4 to 20 kbps per subchannel.

The frequencies of the subcarriers used are communicated by the AOC to all digital receivers, which in turn extract the desired digital signals. The AOC, which is modulated onto one of these subcarriers, periodically informs the receiver which subchannels are being used. The subchannels can each have a number of bits between 1 and 5 per symbol (there are 4000 symbols per second), leading to data rates of 4 to 20 kbps per subchannel.

It should be noted that the transmission system described here does not use the analog FM signal to modulate or demodulate the digital signal; the analog and digital occupy totally separate bands. Thus, the effects of fades upon the analog and digital signals can be analyzed and measured independently and with more assurance than when the two must be separated adaptively.

1.3 AOC Robustness

The AOC can be heavily encoded and protected by a number of means. It is also possible to embed a known, robust signature pattern in the AOC, and then to slide the AOC subcarrier slowly through the band assigned to the digital signal. DSB receivers would then need the ability to scan the band and find the AOC signature (several times for safety). The AOC would continuously transmit essential control information so that receivers would never be more than milliseconds from resetting themselves.

1.4 Complexity/Cost

The AOC allows a number of sophisticated performance- and application-enhancing features, and it may appear that a system using it would have significant cost. However, because the audio speeds are relatively low for modern digital signal processing, it is likely that a low-cost 1-chip realization of a DSB receiver with all the described capabilities could be available in 1994 if design were initiated in mid 1993.

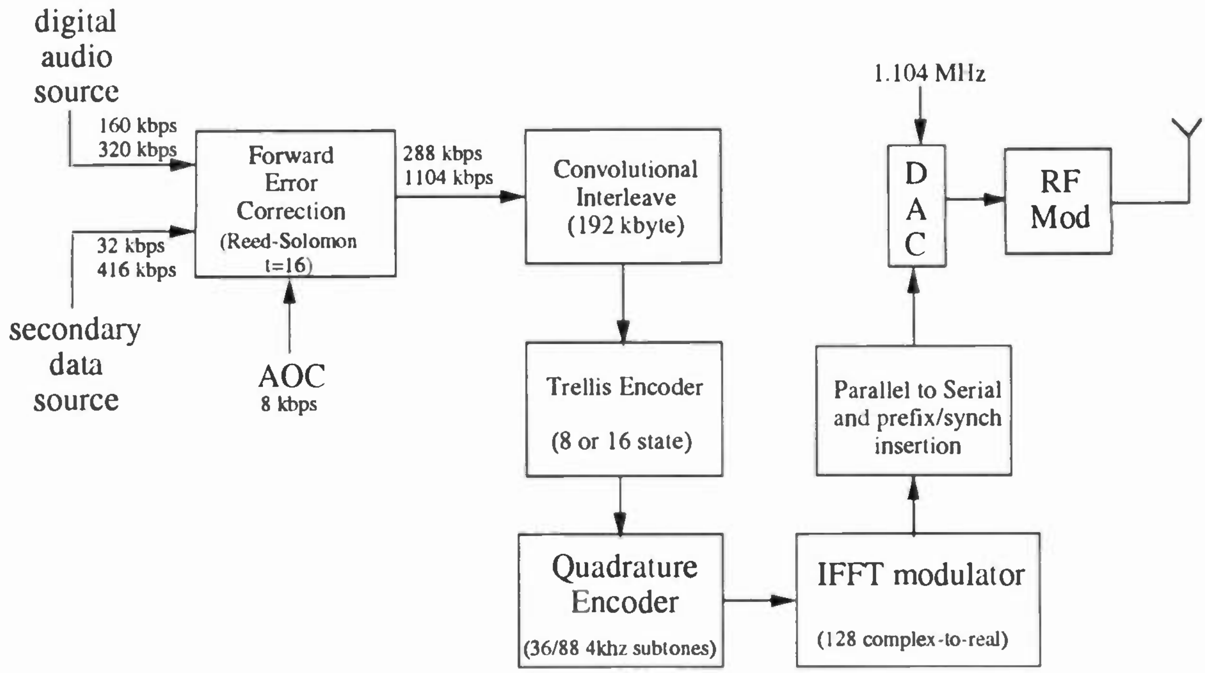

Figure 2 - Block Diagram of DSB Transmitter

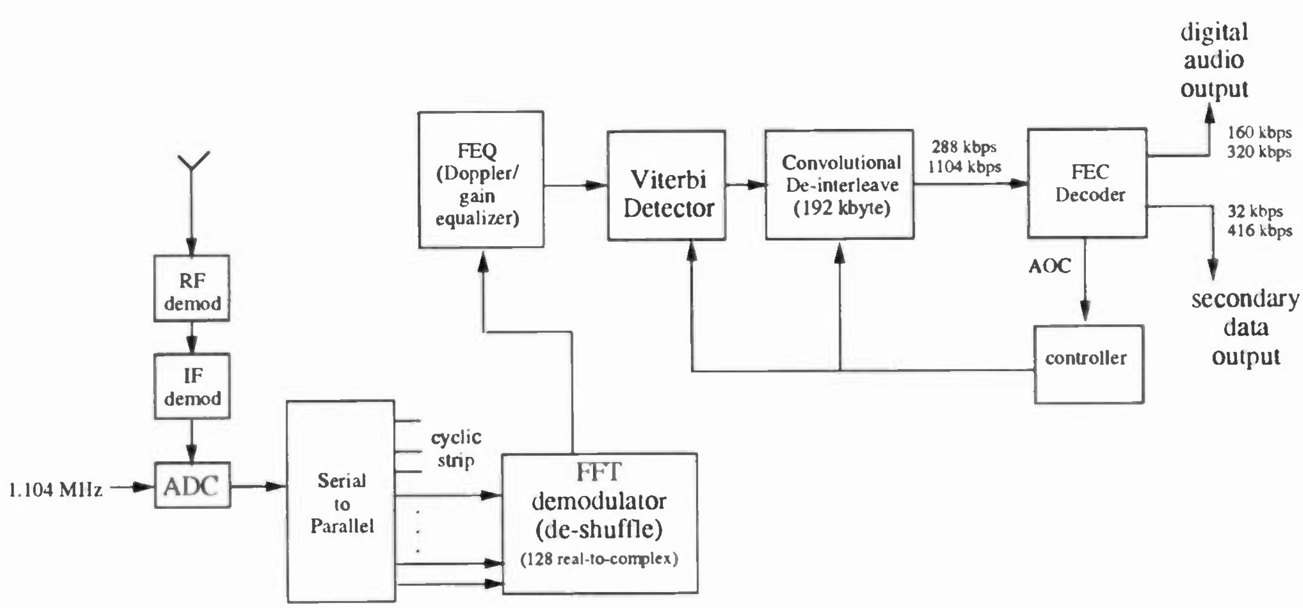

Figure 3 - Block Diagram of DSB Receiver

Table 1 Transmission Commands	
Normal Simulcast	The default distribution corresponding to Figure 1a with the byte after the command byte interpreted as the number of bits used on all non-AOC carriers. (45 tones + AOC in Section 3)
All Digital	The situation in Figure 1b with the byte after the command byte interpreted as the number of bits used on all non-AOC carriers.(90 tones + AOC in Section 3)
Adjacent	The situation in Figure 1c with the byte after the command byte interpreted as the number of bits used on all non-AOC carriers.
Programmed Constant	The next 4 bytes indicate the low and high frequency subchannel indices below carrier and the low and high frequency indices above carrier, respectively. The last frame byte indicates the number of bits used on all non-AOC carriers.
Variable Mode	The last 4 bytes of the next 64 AOC frames specify the subchannel indices and number of symbol bits for two subchannels at a time (thus the variable mode command appears for 64 successive frames when it occurs and it thus takes 1 second to completely reprogram the DSB receiver for unusual applications).
FEC commands	To be determined.
Demultiplex Commands	
DSB	The next byte indicates the lowest subchannel index in the primary digital Sound signal of 192 kbps raw data (the rest of the DSB will be assumed to occupy the next higher subchannels in order until 192 kbps is obtained).
App1- App16	The next two bytes specify the lowest subchannel index and the number of subchannels used by the corresponding application number 1-16.
TBD	Further commands as to be determined.

2. AOC Protocol

The AOC is an 8 kbps data stream, which is implemented on one of the subchannels with 2-bit (sometimes called QPSK) signaling.

The AOC frames are of length 16 ms (62.5 frames/second). Each frame consists of 64 bits or 8 bytes. The first 2 bytes of the frame are the AOC signature bits and are always transmitted in each frame. The next 8 bits are a command byte, so up to 256 possible receiver commands could be implemented, although use of all command combinations is not anticipated The next 5 bytes are sometimes used by various commands to interpret the command byte. Some of the basic commands are listed in Table 1.

From the command set, one can see that the anticipated modes of operation will require only one frame or 16 ms to define because all commands listed can be specified within the eight bytes of one frame. However, some unusual uses could require a few frames to define. The AOC is envisioned as largely static in that the same commands would be repeatedly transmitted unless some new application had just been installed by the broadcaster, in which case the AOC would change just after the application became available.

As with most of the subchannels, fading in the transmission path could cause a loss of the AOC. The default AOC setting will be whatever was sent when the radio receiver first tuned to the broadcast station. A change in application will have to be received for one full second before it is implemented by the receiver to mitigate fading. Thus, a few symbol errors (up to 61 in a row) would be ignored.

3. Implementation

An implementation of a DSB transmitter is shown in Figure 2. Digital audio and/or secondary data at the rates indicated (or other rates) are encoded with a 16-byte error correcting Reed-Solomon code; the AOC data is added and the aggregate data stream is trellis coded. The quadrature encoder and FFT modulate the trellis-encoded symbols onto approximately 37 subcarriers (88 if, as in Figure 1b, there is no analog signal). The modulation is executed with a 128 complex-to-complex IFFT with digital signal processing. The data rates listed could be achieved by modulating 2 bits onto subcarriers in the sidelobes, and up to 5 bits on the subcarriers in the center of the band (because the Signal to Noise plus Interference Ratio is much higher there).

A 16-sample cyclic prefix is added to the IFFT modulator output every 250 microseconds and a 32-sample known synchronization pattern is embedded every 2 ms. The resultant samples are then successively applied to a DAC with sampling rate 1.104 MHz (= 276 x 4 kHz).

The corresponding receiver is shown in Figure 3. After down modulation, the analog signal is sampled at rate 1.104 MHz and stripped of any prefixes or synchronization patterns. The resultant sampled signal is then demodulated by 128-point FFT and compensated for time-varying phase and amplitude distortion by a set of 128 complex 1-tap equalizers (FEQs or frequency-domain equalizers). The update rate of these FEQs (4000 times per second) is sufficiently fast to allow tracking of time-varying gain and phase distortion in the channel, which, at a vehicular speed of 60 mph, varies at a rate of approximately 10 Hz) The trellis-coded symbols are soft-decoded, de-interleaved, and then FEC hard-decoded prior to demultiplexing of the audio, data, and control signals.

4. Conclusion

An auxiliary overhead channel may offer several benefits for digital audio broadcasting. These are programmed migration to new services, the possibility of on-channel simulcast with analog FM, and improved performance and reliability compared to a system without such an AOC.

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AN IN-BAND, ON-CHANNEL FM DIGITAL AUDIO BROADCAST SYSTEM

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ABSTRACT

USA Digital Radio has developed and demonstrated an in-band, on-channel FM digital audio broadcasting (FM-DAB) system. This paper describes the technical approaches taken for extraction of the digital audio signal, non-interference of the underlying FM-DAB signals, and minimization or elimination of multipath.

INTRODUCTION

An in-band, on-channel FM digital audio broadcasting (FM-DAB) system has been developed under the sponsorship of USA Digital Radio and successfully demonstrated in over-the-air testing using the standard FM broadcast signals of radio stations in New Orleans, Louisiana, Urbana, Illinois and Los Angeles, California USA. In developing this system, it was necessary to address three issues: extraction of the digital audio signal from a standard analog FM signal occupying the same spectrum; non-interference of FM-DAB signals with analog FM signals occupying the same frequency channel; and minimization or elimination of the audible effects of multipath. This paper presents an approach to the extraction problem and reviews techniques for minimizing interference and multipath.

EXTRACTION OF DIGITAL SIGNALS

In developing the in-band, on-channel (IBOC) FM-DAB system, it was necessary to extract the digital audio signal from a standard analog FM signal occupying the same spectrum. This section presents an approach to this problem which uses an adaptive transversal filter based on acoustic charge transport (ACT) technology.

Background

Recent developments in semiconductor technology have enabled the practical implementation of a new class of signal-processing components known as adaptive filters. Like all filters, these devices have the ability to separate desired signals from undesired signals; an adaptive filter has the additional capability of automatically changing its response as the characteristics of the desired or undesired signals change.

One form of adaptive filter is the "transversal" filter, which uses successive delay, weighting, and summation operations to pass desired signals while rejecting undesired signals. These filters, also known as "adaptive linear combiners" or "finite-impulse response" (FIR) filters, form the basis of a variety of signal-processing systems which are finding application in the areas of system identification, waveform equalization, signal prediction, and interference cancellation [1].

Until recently, implementation of programmable transversal filters capable of handling high-speed signals has been impractical. While analog components generally have the required processing speed, the critical function of delay has not been realizable in conventional analog circuits. Alternatively, delay is readily available in digital circuits, but the high number of multiply-and-accumulate operations needed by transversal filters limits digital FIR filters to low-bandwidth applications.

High-speed transversal filters are now available through a technology which combines the speed and simplicity of analog components with the programmability and delay capability of digital processing. This technology, known as acoustic charge transport or ACT, has allowed the practical implementation of transversal filters which operate over a frequency range from 500 kHz to 180 MHz, providing several hundred parallel delays over a range of several nanoseconds to several microseconds [2-8]. For applications requiring fast updating of the filter response, ACT filters with update times of less than 100 nsec have been demonstrated. The following sections describe ACT technology, its use in programmable transversal filters, and their application to the signal extraction problem in IBOC FM-DAB systems.

ACT Technology

In its simplest form, an ACT device is an analog delay line in which discrete samples of an input signal are formed as a series of charge packets that propagate in a depleted transport channel induced in a GaAs substrate, as shown in Figure 1. These charge packets are formed and transported by electric fields that arise from piezoelectric coupling to a propagating surface acoustic wave (SAW) which is generated directly on the piezoelectric GaAs substrate. Unlike the operation of SAW devices, the ACT device employs the SAW only as a parametric pump or "clock"; all signal information is contained in the propagating charge packets rather than in the SAW itself. This obviates the deleterious effects of various acoustic phenomena which are observed in SAW devices, since these effects do not directly interact with the charge packets.

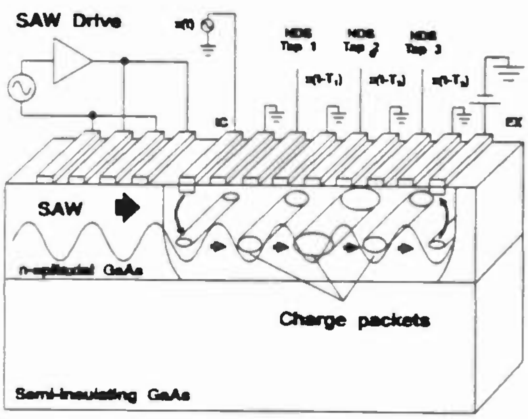

Figure 1. ACT Tapped Delay Line

The charge packets are confined within a transport channel formed in an epitaxial surface layer having a thickness that is an appreciable fraction of the SAW wavelength at the clock frequency. This architecture isolates the charge packets from interfacial traps at the substrate surface and epitaxial layer interface. Thus, ACT devices operate in a deep "buried-channel" mode, and exhibit extremely high transport efficiencies at clock frequencies that are easily achieved in the UHF region (300 to 1000 MHz). The excellent transport efficiency at UHF clock rates forms the basis for a variety of high speed, high performance analog signal processors using ACT technology.

Other technical features that enhance the power of ACT devices in analog signal processors include:

- Intrinsic high performance sampling:

  The phenomenon leading to charge packet formation in the input section of an ACT device results in an intrinsically high performance sampling operation. Effective aperture times have been measured to be less than ten percent of a clock period leading to precise sample formation and the potential for providing frequency conversion directly in the ACT device.

- Non-destructive charge sensing:

  An image charge is induced in arbitrary metal features fabricated within the propagation path that allows the amount of charge in each packet to be sensed without affecting the contents of the packet. This feature enables the realization of high performance tapped delay line structures that form the basis for a rich set of analog signal processors.

- Integration with GaAs ICs:

  Although the fabrication of ACT devices requires a number of unique processing steps, the process is amenable to the integration of GaAs integrated circuit elements to provide interface and control functions. The potential for providing delay and completely integrated RF signal processing functions is unprecedented.

ACT Programmable Transversal Filters

One of the most useful applications of the delay and sensing functions inherent to the ACT device is in the transversal filter. The classical transversal filter is shown in Figure 2. In this device an input signal is processed by passing it through a sequence of delay elements. After each element the signal is sensed, weighted by a predetermined coefficient, and sent to an accumulator. The original signal is passed through another delay element, a different weighting coefficient is applied, and the result is sent to the accumulator. This cycle of delay, sense, weight, and accumulate is repeated in each stage of the filter. The signals from each tap add coherently when the signs of the weighting coefficients match the phase of the input signal and a large signal appears on the output terminal of the filter.

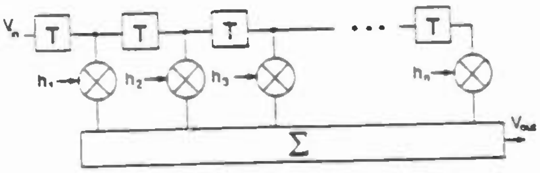

Figure 2. Classical Transversal Filter

The ACT programmable transversal filter uses an ACT tapped delay line to accomplish the delay and sensing operations required by the transversal filter. In this device, a series of non-destructive sensing electrodes are used to sense the input signal as it propagates down the delay line. Each sensing tap is connected to a common summing bus (the accumulator of the transversal filter) through an individual programmable attenuator. The values of the attenuators (the weighting coefficients) are set by a digital controller. All tap weights are stored in random access memory (RAM) which is monolithically integrated with the ACT delay line. A diagram of the ACT programmable transversal filter is shown in Figure 3 and the chip layout in Figure 4.

In filtering applications, the signals to be processed are applied to the analog input port of the ACT delay line, while the tap coefficients are loaded into the on-chip RAM from a digital controller. Standard ACT programmable filters sample at a rate of 360 MHz, have 128 taps spaced by 5.6 nanoseconds, and use 5-bit programmable attenuators for tap weighting. The ACT chip measures approximately 4mm by 6mm, and with control and interface circuitry fits into a 1.25" by 1.25" flatpack.

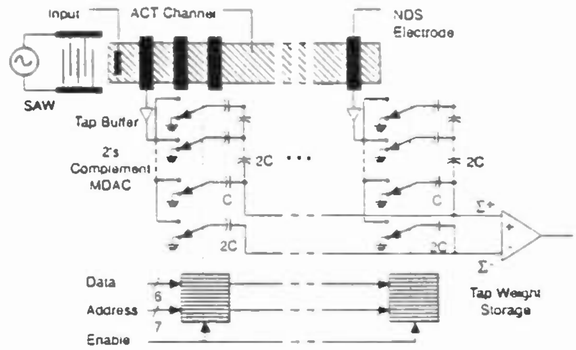

Figure 3. ACT PTF Diagram

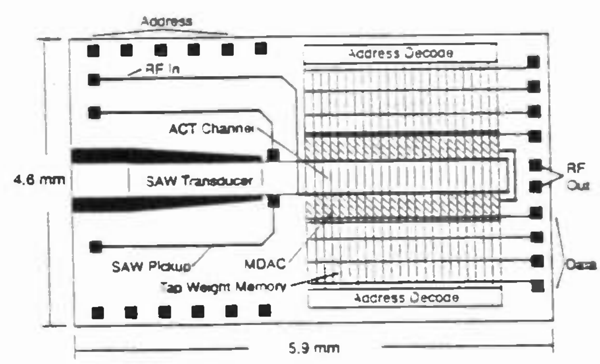

Figure 4. ACT PTF Chip Layout

Adaptive Filtering

Adaptive filters are employed for signal processing applications characterized by temporal variation of one or more parameters of the desired or undesired signals. In these applications, a fixed filter cannot be employed because the filter response cannot be determined a priori. An adaptive filter is therefore required; the parameters of this filter are automatically adjusted by minimizing a selected error measure.

There are two operations fundamental to an adaptive filter: convolution (to perform the filtering function) and filter coefficient update (to optimize the filter parameters). Convolution places the primary computational burden upon any adaptive filter. The strength of the ACT PTF is its effortless ability to perform convolution, overcoming the computational "bottleneck" typical of digital adaptive filters. Its programmable filter coefficients eliminate the tuning and noise difficulties of traditional analog implementations. Simple analog circuits may be employed to perform the adaptive updating for maximum accuracy and flexibility. For applications in which the filter response must be rapidly changed, ACT filter update times of less than 100 nsec have been demonstrated. The ACT PTF therefore combines high-speed capability of analog filters with the flexibility of digital processing. The operating frequencies of the ACT-based PTF allow it to be used directly in the RF spectrum. For the IBOC FM-DAB application, it is most convenient to operate the adaptive filter directly at the IF stage of the receiver.

Analog FM Broadcast Signal Filtering

In order to extract a low-level digital signal from beneath a standard analog FM signal, an ACT-based adaptive filter has been developed which operates at the standard IF frequency for FM receivers. The specifications for this filter are shown in Table 1.

Table 1

Parameter	Specification
Center frequency	10.7 MHz
Frequency deviation	+/-82 kHz
Modulating signal	15 kHz stereo
Filter update rate	< 100 nsec
Analog signal suppression.	>35 dB
Size (chip)	4 x 6 mm
Size (hybrid)	1.25" x 1.25"
Power consumption	< 3 W

Filters meeting these specifications have been designed, fabricated, and tested. These filters have been used in the over-the-air IBOC FM-DAB system demonstrations conducted in New Orleans, LA, Urbana, IL and Los Angeles, CA USA. The measured performance of these filters is presented in the following section.

Demonstrated Results

The spectrum of a standard FM broadcast taken from the IF stage of a commercial receiver and measured on a spectrum analyzer is shown in Figure 5a. The peak amplitude is seen to be approximately -15 dBm, and the -30 dB bandwidth is seen to be approximately 200 kHz. The same signal is shown in Figure 5b after cancellation by a standard ACT-based adaptive filter. In this case, peak amplitudes of approximately -50 dBm are measured, indicating that a suppression of at least -35 dB has been achieved. The response of this filter is updated in less than 100 nsec. Since this data was taken, an additional 10 dB of suppression has been measured using ACT-based adaptive filters custom-designed for IBOC FM-DAB applications.

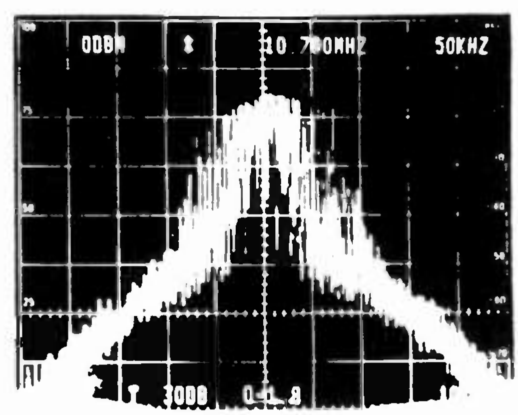

Figure 5A. Spectrum of FM Signal

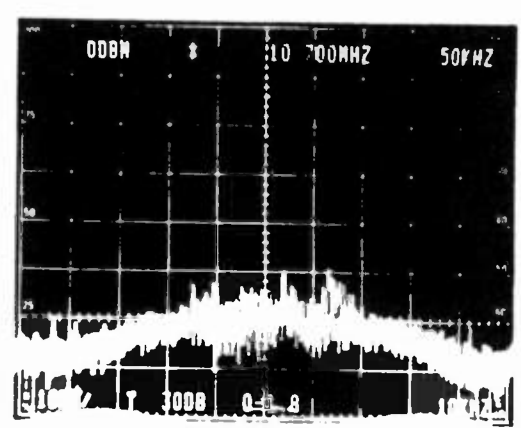

Figure 5b. Spectrum of Cancelled Signal

The ability to recover a digital signal occupying the same spectrum as the analog FM signal is demonstrated in Figure 6. The top trace in this figure shows the digital signal prior to modulation and transmission. The bottom trace shows the recovered digital signal after cancellation of the analog FM signal and demodulation of the digital signal. During over-the-air testing, digital signals with data rates of up to 192 kBits/sec have been successfully extracted from as much as 45 dB below standard analog FM signals occupying the same frequency allocation.

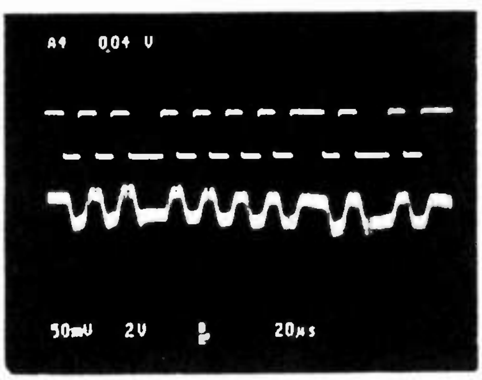

Figure 6. Underlying Digital Transmission, Before Modulation and After Cancellation and Demodulation

NON-INTERFERENCE OF IBOC DAB SIGNALS WITH ANALOG FM SIGNALS

By definition, an in-band, on-channel FM-DAB signal simultaneously occupies the same frequency channel as a conventional analog FM broadcast signal. The characteristics of the digital signal must therefore be designed to prevent degradation of the analog signal. The most obvious approach to minimizing interference is to reduce the amplitude of the digital signal relative to the analog signal. Of course, the amplitude of the digital signal cannot be made arbitrarily small, since interference from the analog signal and thermal noise will ultimately degrade the digital signal. Once the digital signal amplitude has been reduced to the smallest possible level which maintains the desired bit error rate over the desired coverage area, another technique must be used to ensure non-interference with the analog signal. One such technique is to design the frequency-domain characteristics of the digital signal such that it is orthogonal to the analog signal. This approach will be described below, after a brief discussion of the amplitude-reduction approach.

IBOC DAB Signal Levels

Digital signals occupying the same frequency channel as analog FM signals add a random-noise component to the received signal. Recent experimental evidence indicates that a 50-dB audio (post-detection) signal-to-noise ratio (SNR) requires between 30 and 50 dB of "protection" against digital quadrature phase-shift keyed (QPSK) and quadrature amplitude modulated (QAM) signals [9]. The wide variation in these protection ratios arises from differing test methods; using a CCIR-recommended quasi-peak detection approach gives protection ratios of 38.5 to 48.5 dB to maintain 50-dB audio SNR in the presence of co-channel digital 256-QAM and QPSK signals. An alternative approach using an RMS detection to measure "unweighted" SNR yields protection ratios of 30 to 32 dB for co-channel 256-QAM digital signals, although significant audio degradation was observed at these levels.

IBOC DAB Signal Orthogonality

Additional suppression of interference is achieved by modulating the digital signal in a way that ensures that it is orthogonal to the analog signal. One method of achieving this orthogonality is to design the digital signal spectrum such that it is never superimposed directly on the analog signal spectrum. While frequency separation may not seem feasible for in-band, on-channel systems, it is accomplished in systems which employ frequency sliding of the digital signal. Frequency sliding is a constant offset between the analog-FM and IBOC DAB signals maintained by modulating the center frequencies of the digital carriers for each subchannel by the FM program. This allows the correlation between the digital and the analog signals to be minimized, which also minimizes the mutual interference between the signals.

Of course, practical system implementations cannot be expected to maintain perfect orthogonality, and any correlation between the analog and digital signals will result in some amount of mutual interference between the signals. The amount of interference will depend on the ability to prevent any overlap between the analog signal spectrum and the digital signal spectrum. By proper design and implementation of the digital waveform, clear-channel (multipath free) interference suppression of 10 to 20 dB is readily accomplished. Thus the protection ratio of 40 to 50 dB indicated by the testing performed in [9] is achieved by transmitting the digital signals approximately 30 dB below the analog carrier level.

Minimization of the correlation between the analog and digital signals in a clear (multipath-free) channel does not guarantee minimal interference in the presence of multipath. This is because the delayed-path signals will be at a different frequency than the direct-path signal due to the modulation of the digital carriers by the FM program. If the direct-path signal is made orthogonal to the analog signal, the delayed-path signal will not be perfectly orthogonal, and the amount of interference between the signals will depend on the frequency difference (which is proportional to the delay time of the echo and FM rate). To minimize interference in this case, the separation between the analog signal spectrum and the digital signal spectrum must be made sufficiently large to maintain orthogonality even in high-multipath environments. For multipath delay spreads of 1 to 5 microseconds, this separation must be between 5 and 30 kHz for 110 percent modulation. For extreme multipath environments with sufficient delay spread to cause loss of orthogonality between the analog and digital signals, the analog FM signal will be degraded by the multipath to the point at which the decrease in signal-to-noise ratio caused by the digital signal is expected to be imperceptible at the audio output of the receiver.

Measured Performance

Non-interference of an in-band, on-channel FM-DAB signal with conventional analog-FM reception has been demonstrated during over-the-air testing performed with station WWNO in New Orleans, Louisiana, station WILL in Urbana, Illinois and station KTWV in Los Angeles, California. The analog exciters, transmitters, and antennas of all three stations were used without modification; the digital signals were injected into the transmission line between the high-power analog transmitter and the antenna. The interference testing consisted of three components:

1. S/N measurement at the transmitter site
2. Audio tests at fixed locations and in a mobile platform
3. Listener response during multi-week operation.

The first interference test component consisted of direct measurements of the transmitted signal-to-noise ratio (S/N) at the transmitter site. Prior to injection of the digital signal, measurements made at station WILL-FM in Urbana, Illinois in September 1992 yielded a S/N of approximately 60 dB; the injection of a frequency-sliding 192-kBit/sec digital signal with a power level of approximately -30 dBc decreased the measured S/N by approximately 0.5 dB, to 59.5 dB.

After measuring the S/N at the transmitter site, a standard commercial automobile FM receiver was used to attempt to detect the presence of the digital signal in the audio output of the receiver. For these tests, the digital transmitter power was cycled on and off while two engineers attempted to discern any change in the quality of the audio. During both mobile and stationary testing, the presence of the digital signal was never detected. In some locations in which severe multipath caused almost total loss of the analog signal, the digital signal remained undetectable.

The third component to the interference testing took place over approximately four weeks in Urbana, Illinois, where an in-band, on-channel DAB was broadcast over WILL-FM for a period of approximately four weeks. During this time, no listener complaints correlated to the IBOC DAB testing were reported.

MULTIPATH MITIGATION TECHNIQUES

In addition to CD quality stereo programming and improved data services, digital audio broadcast promises to mitigate the adverse impact of multipath. In-band, on-channel (IBOC) FM-DAB systems deliver CD-quality audio within existing spectral channels while not interfering with existing FM broadcast reception. Most measures proposed for mitigating multipath involve the use of new spectrum. This section explores those measures which are being taken to mitigate multipath for the in-band, on-channel DAB system now under development.

Multipath is the time domain phenomenon wherein successively delayed versions of a broadcast signal arrive at the receiver simultaneously. Multipath is typically random and time variant for a moving vehicle. A multipath channel time response has an associated frequency response. Multipath is usually characterized in the frequency domain in terms of amplitude fade depth, spatial and temporal correlation of fade depths, and frequency coherency which relates to fade bandwidth. Techniques employed for mitigating multipath include:

- spread spectrum modulation,
- data encoding,
- frequency division multiplexing,
- adaptive channel equalization, and
- time, frequency and spatial diversity.

Spatial diversity in the form of multiple antennas has been shown to be helpful in improving FM reception in automobiles; however, due to practical and aesthetic considerations, the use of multiple antennas has not been accepted by the FM radio industry and is not considered a part of the solution for multipath mitigation. Spread spectrum has been clearly shown to alleviate multipath, but bandwidth requirements for applying spread spectrum techniques to IBOC DAB are not consistent with existing spectral occupancy for FM.

Frequency diversity becomes effective against multipath as spectral separation employed begins to exceed multipath coherence bandwidths. Urban FM multipath is thought to have coherence bandwidths in the 30 to 300 kHz range and is thought to be resistant to in-band on-channel frequency diversity techniques; however, FCC 73.317 defines the spectral occupancy for commercial FM in the United States over a 1.2 MHz bandwidth. Compliance with FCC 73.317 allows the power within 480 kHz of this bandwidth to reach -25 dBc. Using some of this power for IBOC DAB allows for a level of frequency diversity which is exploited towards the mitigation of multipath.

Adaptive channel equalization has been shown to improve multipath reception in radio systems, but the rapidly varying nature of multipath in automobiles precludes the use of conventional adaptive equalization techniques. A recently developed measure which compensates for multiple delayed echoes is being taken to effectively equalize IBOC DAB in real time. Frequency division multiplexing, data encoding and time diversity complement frequency diversity measures and recently developed equalization techniques for a comprehensive IBOC FM-DAB system with surprisingly high multipath resistance.

Multipath Mitigation through Modulation

Three measures inherent to the modulation are employed to mitigate multipath in the IBOC FM DAB system now under development: a frequency slide technique, frequency division multiplexing, and a recently developed equalization technique.

"Frequency sliding" is a modulation technique in which the carrier frequencies of a series of digital subchannels are modulated by the FM program. This has the effect of producing a constant frequency offset between the analog-FM carrier and the IBOC digital signals. The primary motivation for IBOC DAB frequency slide is that sliding the DAB carrier frequencies in synchronization with the instantaneous FM signal frequency may be used to make conventional FM detection techniques insensitive to IBOC DAB. The added benefit of frequency slide is multipath mitigation. Frequency slide increases the effective IBOC DAB bandwidth for multipath mitigation without increasing the IBOC DAB noise bandwidth. Frequency slide contributes a level of effective frequency diversity against multipath.

Frequency division multiplexing is a common practice for mitigating multipath. The time domain advantage of frequency division multiplexing is the reduction of intersymbol interference (ISI) in each subchannel due to multipath with respect to the ISI which would otherwise be seen by the proportionally shorter duration symbols on a single carrier. The frequency domain advantage of frequency division multiplexing is the isolation of the effects of narrowband fading to a fraction of the subchannels. The errors induced on the affected subchannels are recovered through data decoding.

The recently developed equalization

technique is used to compensate for nonuniform phase distortion induced by multipath across the band. This measure allows for the coherent contribution of delayed signal components to the digital demodulation process. All the delayed signal components contribute coherently to the demodulation of each data symbol. The processing gains are analogous to those of an ideal channel equalizer with no adaptation time.

Multipath Mitigation through Coding

Data encoding and error correction are the subject of substantial research efforts for a variety of communications and data storage applications, and the power and efficiency of these techniques have increased significantly as a result of these efforts. Three measures inherent to the data encoding technique are used to mitigate the effects of multipath in the IBOC FM-DAB system now under development: block coding, convolutional coding, and data interleaving.

Block coding is used to detect and correct errors. A portion of the errors due to narrowband fades or to temporary fades in a moving vehicle may be detected and corrected through block coding. Block coding is also referred to as "error detection and correction."

Convolutional coding provides processing gain against losses of signal level through soft decision Viterbi decoding. Convolutional coding also distributes information across multiple carriers which adds a level of effective frequency diversity to the modulation. Soft decision Viterbi decoding essentially gives the decoder the demodulation information as well as subchannel reliability information. Data is decoded according to a set of relative subchannel confidence metrics.

Data interleaving is used to distribute burst errors between levels of coding so as to make burst errors appear random. Although convolutional encoding adds processing gain to the demodulation process, errors which do propagate through soft decision Viterbi decoding are usually bursty in nature. Interleaving spreads out burst errors in time so as to enable correction by the block decoder.

Combined Advantages of Multiple Multipath Mitigation Techniques

The primary effects of multipath in an IBOC DAB system and the techniques used to mitigate these effects are summarized in Table 2 (after [10]). Inter-symbol interference is caused by strong echoes with long delay times; most experimental evidence suggests that delay spreads of 1 to 5 microseconds are common [10-13], although some measurements indicate that echoes of 15 or even 30 microseconds may occur under some circumstances [14]. As described above, one common technique used to combat this effect is frequency-domain multiplexing (FDM), in which a high data-rate signal is divided into a number of lower data-rate signals which are then transmitted in lower-bandwidth subchannels. The symbol time in each subchannel is made longer than the longest expected delay; this suggests subchannel data rates of less than 33 kHz (1/30 microsec) should be used for the conditions described above.

FDM has the effect of confining the impact of long-delay multipath to a small number of subchannels; combining FDM with frequency sliding and forward error correction provides significant system robustness against multipath. The primary benefit of frequency sliding in this regard is to minimize the effect of multipath on any one channel by moving the subchannels through the affected frequency range. Forward error correction may then be applied to restore the data lost in the affected subchannels. Additional protection against inter-symbol interference is provided by the channel equalization techniques described above.

Table 2

Multipath Effect	Mitigation Technique
Inter-symbol interference (caused by strong echoes with long delays)	Frequency-domain MUXing Frequency sliding Forward error correction Channel equalization
Amplitude fading (bandwidth inversely proportional to delay spread)	Frequency diversity -Subchannel spreading -Frequency sliding Coding and interleaving

These techniques offer some amount of protection against the amplitude fading effects of multipath, although the amount of protection depends on the characteristics of the fading. In particular, the coherence bandwidth of the fading is a critical factor in determining the effectiveness of these mitigation techniques. Measurements made at frequencies near 900 MHz have suggested that typical rural and urban multipath will cause fading with coherence bandwidths of

$$B = 60/D \quad \text{to} \quad 90/D$$

where B is the 90% coherence bandwidth in kHz and D is the delay spread in microseconds [15,16]. For delay spreads of 1 to 5 microseconds, this gives coherence bandwidths of less than 12 to 90 kHz. In these cases, amplitude fading will affect a limited number of subchannels, and the frequency diversity techniques described above will be effective.

Statistics of the amplitude fading depth may be combined with the coherence bandwidth measurements to estimate the expected signal attenuation of an IBOC DAB signal caused by multipath at various locations. Experimental evidence indicates that the small-sector field strength obeys the Rayleigh distribution [16]; this suggests that 10% of locations will experience fading of at least 8 dB relative to local average field strength and 1% of locations will experience fading of at least 18 dB. It is important to note, however that this fading applies to narrowband signals, and the correlation between these fading statistics and coherence bandwidth must also be considered. Assuming that fading depth and coherence bandwidth are statistically independent, we may estimate the probability that a given location will experience a specified level of fading over a specified bandwidth by multiplying the probabilities of each occurrence. Since coherence-bandwidth measurements indicate that fades with 300-kHz bandwidth occur in less than 10% of the locations surveyed [15], it is reasonable to conclude that less than 1% of locations will experience 8-dB fades over 300 kHz and 0.1% will experience 18-dB fades over 300 kHz. In these cases, multipath mitigation must be accomplished through data coding, interleaving, and error correction. An important factor in determining the effectiveness of these techniques is the time duration of the multipath-induced fading.

For mobile receivers, the amount of time the receiving antenna spends in the fading region is determined by the spatial correlation of amplitude fading and the velocity of the platform. Measurements indicate that spatial correlation distances from 7 inches to 2 feet are common [17]; theoretical analysis predicts correlation distances of up to 7 feet [16]. For a vehicle travelling at 20 miles per hour, these multipath correlation distances would require data coding and correction which can handle reduced-quality data for periods of tens to hundreds of milliseconds. This is within the capabilities of recently developed burst-error detection and correction techniques.

CONCLUSIONS FOR DIGITAL AUDIO BROADCASTING

Three issues have been addressed in the development and demonstration of an in-band, on-channel FM-DAB system: extraction of the digital signal, non-interference of the FM-DAB signals with analog FM signals and multipath mitigation. The conclusions reached for each of these areas are:

1. Adaptive signal processing techniques have been demonstrated which allow cancellation of the analog FM signal by more than 35 dB while allowing the underlying digital signal to pass through.

2. Simultaneous transmission of in-band, on-channel FM-DAB signals which do not interfere with conventional analog-FM signals has been demonstrated.

3. Multipath mitigation techniques exist which provide significant robustness against multipath while occupying a spectrum no wider than existing channels and not interfering with existing FM.

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UHF TRANSMISSION

Monday, April 19, 1993

Moderator:
Jerry Whitaker, Technical Writer, Beaverton, Oregon

THE EXPERIENCE OF HIGH-POWER UHF TETRODES
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Thomson Tubes Electroniques
Thonon Les Bains, France

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Harris Corporation, Harris Allied Division
Quincy, Illinois

*Paper not available at the time of publication.

THE EXPERIENCE OF HIGH-POWER UHF TETRODES

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Abstract

The 90's will see the implementation of HDTV and the rapid penetration of digital transmission in TV broadcasting systems. Based on several years of experience with high-power transmitters, the paper will describe the performances of the tetrodes in present transmitters. Then it will be shown how these performances are consistent with digital broadcasting. The reliability and cost saving aspect will be examined in conclusion.

I - INTRODUCTION

30 years ago the first 10 kW tetrode was installed in a UHF transmitter. 10 years ago the first 20 kW tetrode was installed in a separate amplification UHF transmitter. For 4 years already tetrodes have been running at 25 kW in combined amplification UHF transmitters. **In 1992, the first 30 kW combined amplification UHF transmitter appeared featuring the advantages of a tetrode.** The characteristics of these components, improved over the last years, seem to be well adapted to the preliminary specifications of digital HDTV. This paper will describe the tetrode characteristics with regard to their use in digital transmission.

II - TETRODE PERFORMANCES IN STANDARD TV TRANSMITTERS

For several years, many transmitters using tetrodes have been installed all over the world with peak powers up to 30 kW. Over 10 kW, these transmitters use water cooled tetrodes. This technology makes possible a large increase in anode dissipation reaching power density over 1 kW/cm² on the anode with the Hypervapotron*) system. At these power levels, **TTE patents concerning anode dissipation and screen grid base cooling eliminate any gain variation or phase shift during the whole life of the components.** Thus no corrections are necessary in the transmitter during this period. The two most powerful UHF tetrodes TH 582 and TH 563 employ these technologies.

*) TTE patent

The TH 582 mainly used in the resonant cavity TH 18582 delivers 10 kW in common amplification transmitters and 20 kW over separate amplification transmitters in the whole range of UHF frequencies. The operating conditions of this tube at 10,5 kW common amplification are summarized in Table 1.

Frequency	700 MHz
-1 dB bandwidth	12.5 MHz
Operating heater voltage	3.9 V
Anode voltage	5.5 kV
Control-grid bias voltage	-90 V
Screen-grid voltage	600 V
Anode current at zero signal	1.5 A
Anode current	3.45 A
Control-grid current	20 mA
Screen-grid current	50 mA
Gain	15 dB
Peak-of-sync output power	10.5 kW
Sound output power	1.05 kW
Average output power at black level + sound	7.0 kW
Average input power	220 W
Intermodulation ratio	48 dB

TABLE 1
TH 582 10,5 kW common amplification
(CCIR G Standard)

The TH 563 was first used at 20 and 25 kW common amplification in Acrodyne transmitters. The very good performance of the tube under these conditions pushed other transmitter manufacturers to use this tetrode and led us to increase the output power of the tube. Now three manufacturers in the USA and Europe use the TH 563 at power level up to 30 kW in common amplification.
The operating conditions of this tube at **31.5 kW common amplification** in the cavity TH 18550 are summarized in Table 2.

Frequency	700 MHz
-1 dB bandwidth	12 MHz
Operating heater voltage	4.9 V
Anode voltage	8.5 kV
Control-grid bias voltage	-113 V
Screen-grid voltage	800 V
Anode current at zero signal	2 A
Anode current	6.45 A
Control-grid current	150 mA
Screen-grid current	130 mA
Gain	14.5 dB
Peak-of-sync output power	31.5 kW
Sound output power	3.15 kW
Average output power	21 kW
Third order intermodulation ratio	48 dB
Average input power	0.74 kW

TABLE 2
TH 563 31.5 kW common amplification
(CCIRG standard)

In order to judge the performance of components, it is necessary to consider all the quality measurements made on the tube and circuits under well defined operating conditions. In fact, every operating condition is the result of a trade off between different parameters. In the optimization of these conditions, the efficiency, the gain or the TV signal quality without corrections may be enhanced. While keeping good values for gain and efficiency, it is important to minimize the correction to be applied on the TV signal thus decreasing the complexity of the transmitter and increasing its reliability. **The excellent linearity of tetrodes, mainly due to the use of pyrolytic graphite grids,** leads to the following performances without correction :

* differential gain illustrated in Fig. 1 shows a maximum value of 15 %
* differential phase illustrated in Fig. 2 is 3°

* compression on the synchronization pulse is 5 %, see Fig. 3
* ICPM shown in Fig. 4 : 3° in the output for 1° in the input
* bandwidth is 12 MHz at -1 db as shown in Fig. 5
* intermodulation, the best illustration of the linearity of this tube, is measured at -50 dB see Fig. 6

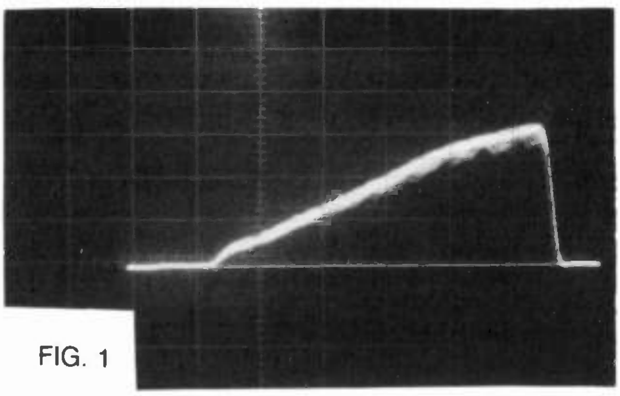
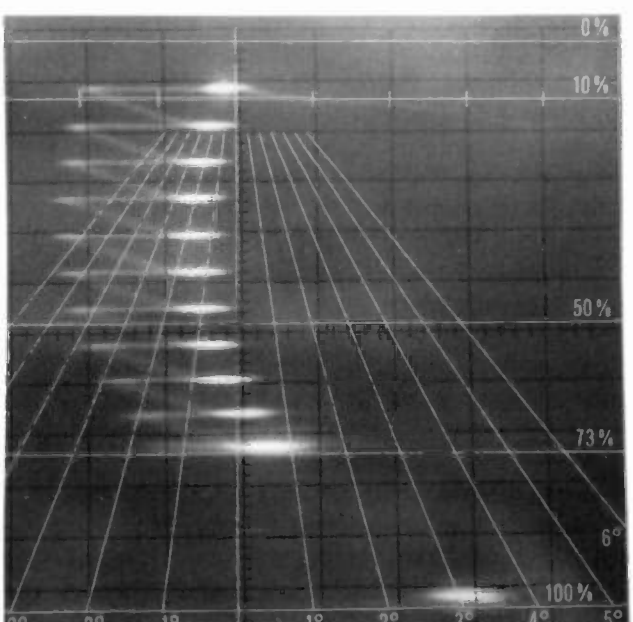


FIG. 1

TH 563 at 31.5 kW Diff. gain 5% / Div.



TH 563 at 31.5 kW ICPM

FIG. 4

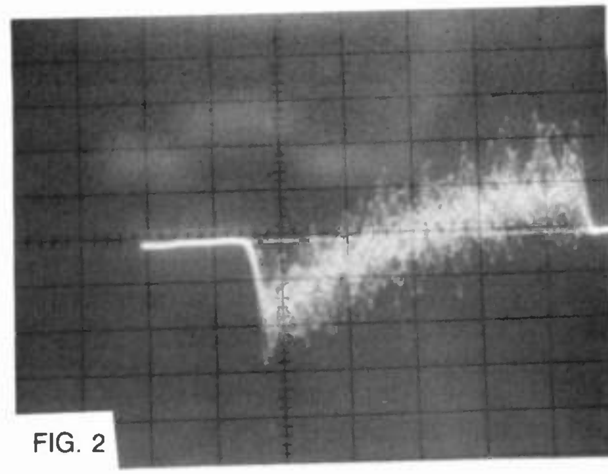


FIG. 2

TH 563 at 31.5 kW Diff. phase 2°/ Div.

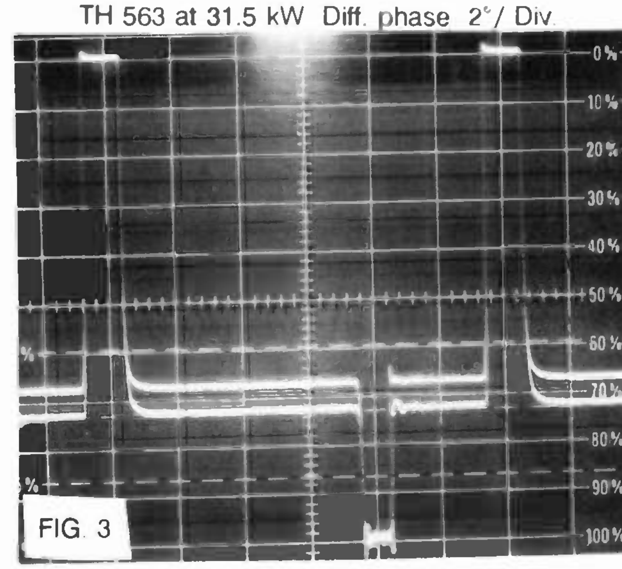


FIG. 3

TH 563 at 31.5 kW Synchronization pulse

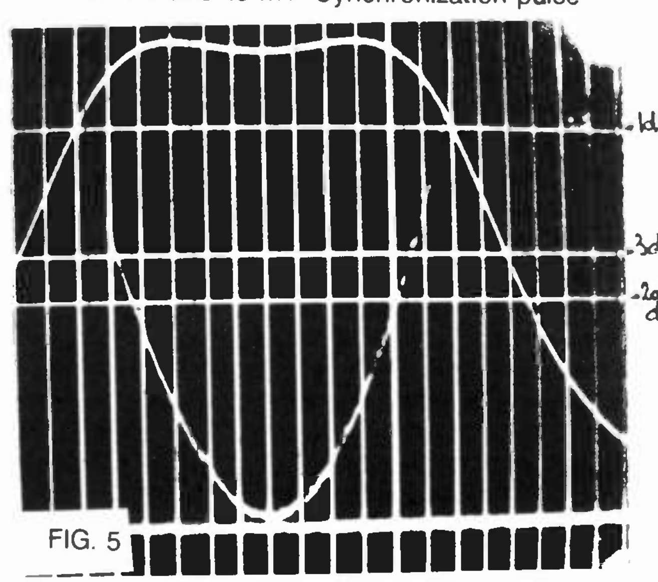


FIG. 5

TH 563 Bandwidth at 700 MHz

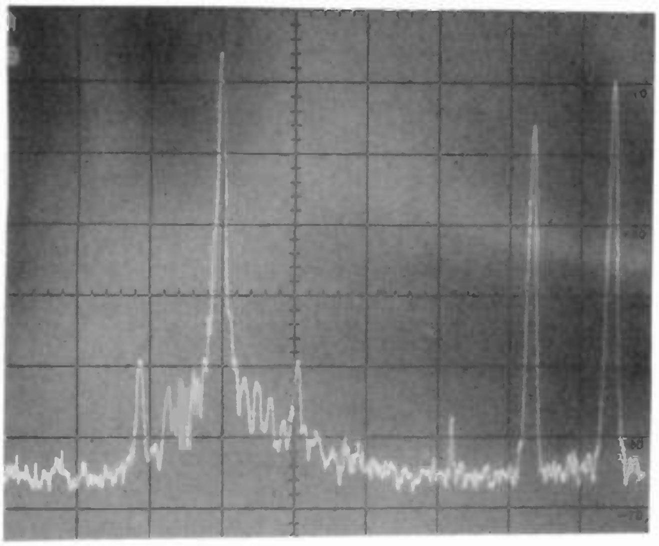


FIG. 6

TH 563 at 31.5 kW Intermodul. 10 dB/Div.

The power reserve and linearity of the TH 563 also appear clearly in the case of separate amplification. Under these conditions, the tube is able to deliver up to **44 kW peak-of-sync power** with very good gain and efficiency performances as shown in Table 3.

Frequency	700 MHz
-1 dB bandwidth	12 MHz
Operating heater voltage	4.9 V
Anode voltage	9 kV
Control-grid bias voltage	-114 V
Screen grid voltage	800 V
Anode current at zero signal	2 A
Anode current	6.75 A
Control-grid current	90 mA
Screen-grid current	70 mA
Gain	14.7 dB
Peak-of-sync output power	44 kW
Average output power at black level	25 kW
Average input power at black level	0.84 kW

TABLE 3
TH 563 44 kW visual only
(CCIRG standard)

As for common amplification, the operating conditions have to be read when no corrections have been made :

- compression of the synchronization pulse (Fig. 7) : 6 %
- differential gain (Fig. 8) shows a maximum value of 8 %
- differential phase (Fig. 9) is 6°

In addition to these results in TV operation, the safety margins of the TH 563 have been tested in pulse conditions at 110 kW and 500 MHz. A CW test has also been made at 45 kW at the same frequency.

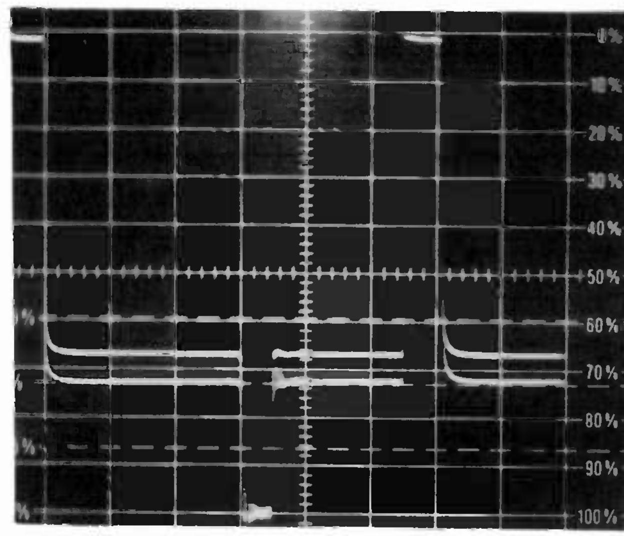


FIG. 7

TH 563 at 44 kW Synchron. pulse

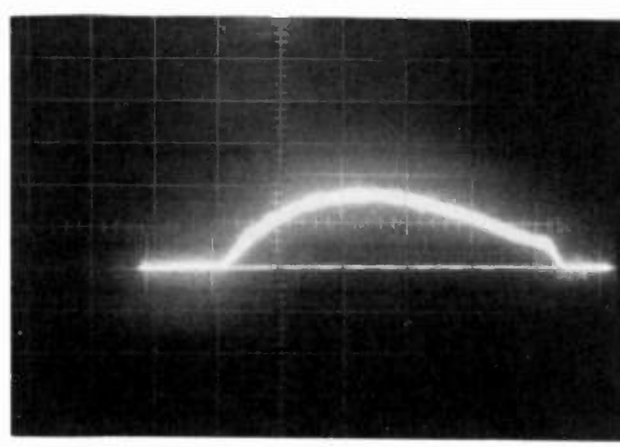


FIG. 8

TH 563 at 44 kW Diff. gain 5 %/Div.

III - DIGITAL TRANSMISSION THE INTEREST OF THE TETRODES

Since the final specifications for HDTV have not yet been decided, it is very difficult to foresee the future, however, looking to the different systems in competition for digital transmission, some common characteristics appear :

- The peak power of the transmitter could be in range of 30 to 60 kW
- The ratio between peak power and average power could be around 7 to 10 db. Then the average power would be in the range of 3 to 12 kW.
- The necessary flat bandwidth will be 6 MHz in the USA.
- The system will require a very high level of linearity in amplitude and in phase in order to minimize the BER

As we saw previously, the power range of tetrodes is fully consistent with HDTV requirements. As a matter of fact the peak to average power ratio being much higher than for standard transmission, it thus becomes possible to increase the peakpower while keeping the meanpower below current proven performances. Then the tetrodes will have reduced RF losses and so reduced internal temperature, thus leading to even better life-time . This point will be discussed later.

In order to show the performance of tetrodes under HDTV conditions, we made some measurements at 31.5 kW peak-power assuming 7 dB between peak and mean power. Tests were carried out at 706 MHz.
The bandwidth obtained with cavity TH 18550 and the tetrode TH 563 was measured at low and full power. These measurements show that even without corrections, the bandwidth' degradation remains below 1 dB within 6 MHz.

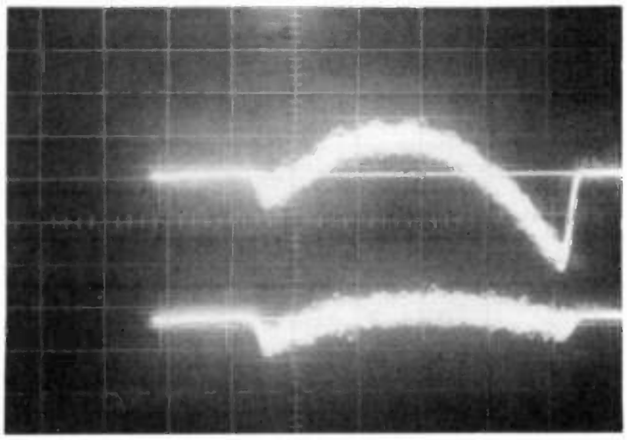


FIG. 9

TH 563 at 44 kW Diff. phase 2°/Div.

We saw in the previous paragraph that the differential phase at 31.5 kW was 6°. This result leads us to think that even at higher power this very low phase shift is one of the most attractive characteristic of tetrodes when looking at QAM system.
In fact, phase distorsion will be the most sensitive point in this case.

Intermodulation measurements have been made under different conditions, showing excellent linearity of the tetrode. These two tone measurements were carried out at -6 db, -9.5 dB and -12 dB for a gap of 2 MHz and 0.5 MHz and at a central frequency of 700 MHz. All these measurements have been carried out without corrections. The results at 26.5 kW are shown in Fig. 10-11-12-13.
The intermodulation never exceeds -40 dB in the worst case. At 31.5 kW, the results are shown in Fig. 14-15-16-17.
In this case, the intermodulation is better than -37 dB showing the great interest of tetrodes for linear applications.

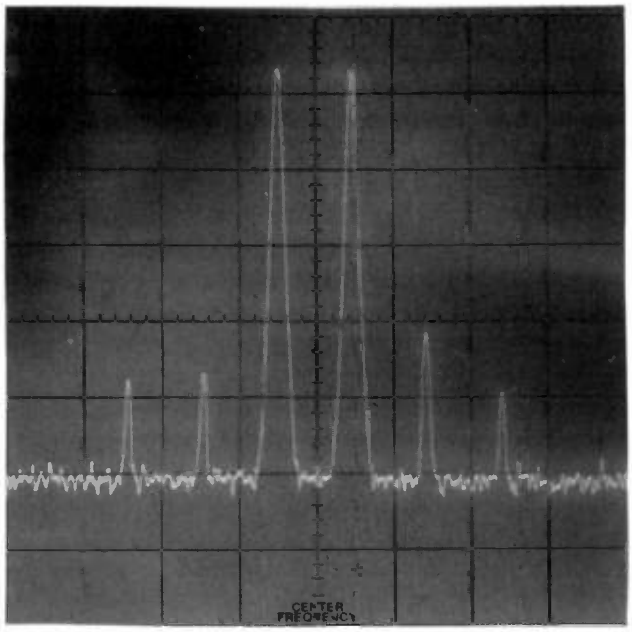


FIG. 10

TH 563 at 26.5 kW -6 dB Δf 2 MHz

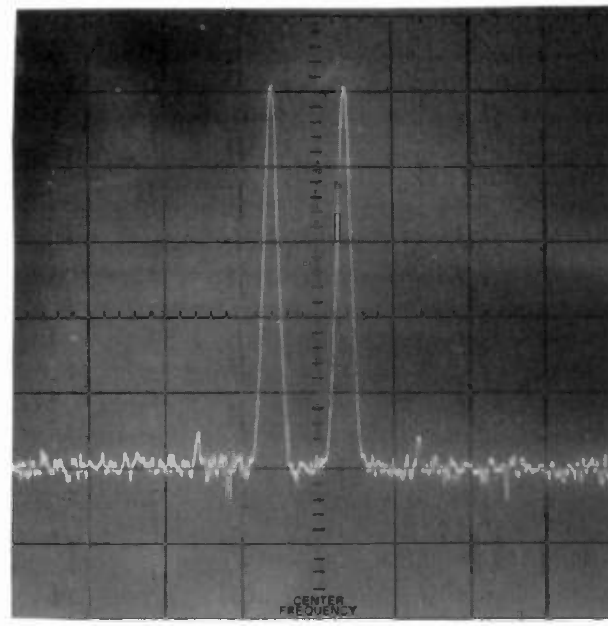


FIG. 12

TH 563 at 26.5 kW -9,5 dB Δf 2 MHz

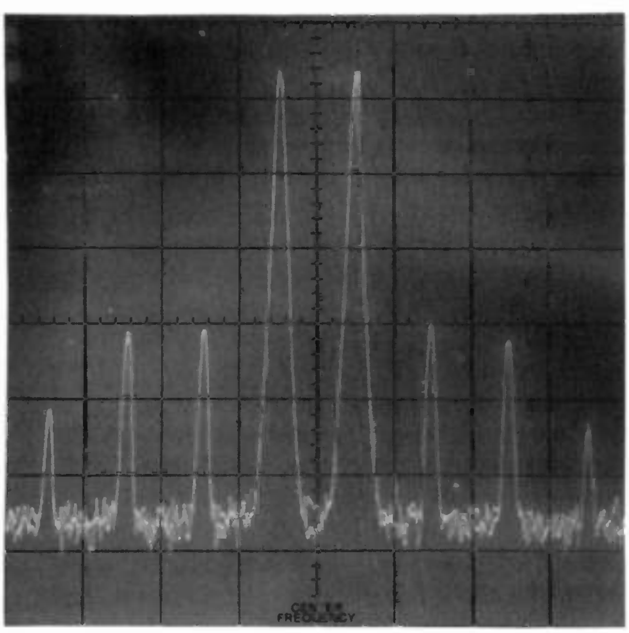


FIG. 11

TH 563 at 26.5 kW -6 dB Δf 0.5 MHz

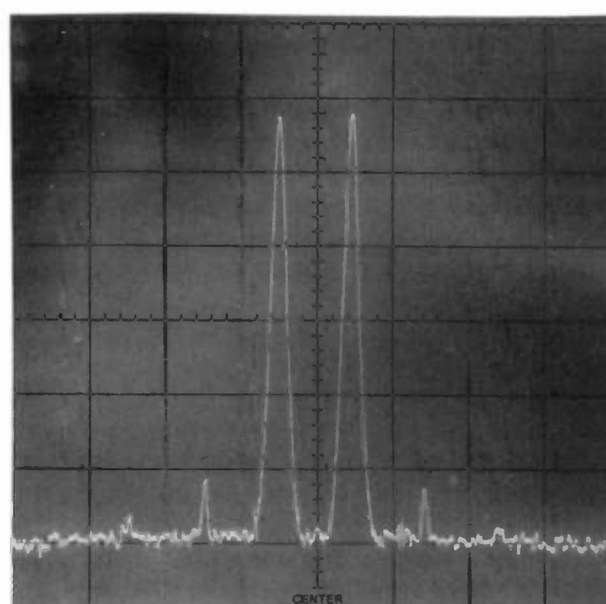


FIG. 13

TH 563 at 26.5 kW -12 dB Δf 2 MHz

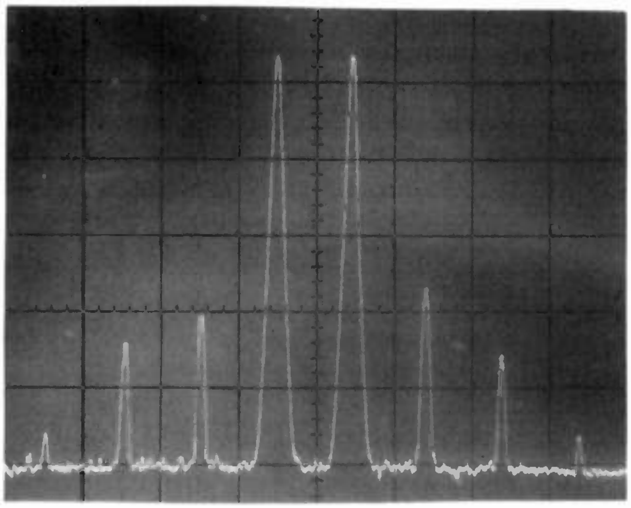


FIG. 14

TH 563 at 31.5 kW -6 dB Δf 2 MHz

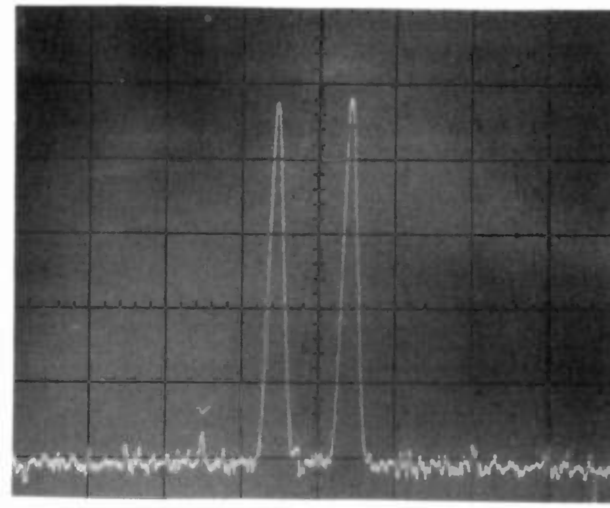


FIG. 16

TH 563 at 31.5 kW -9,5 dB Δf 2 MHz

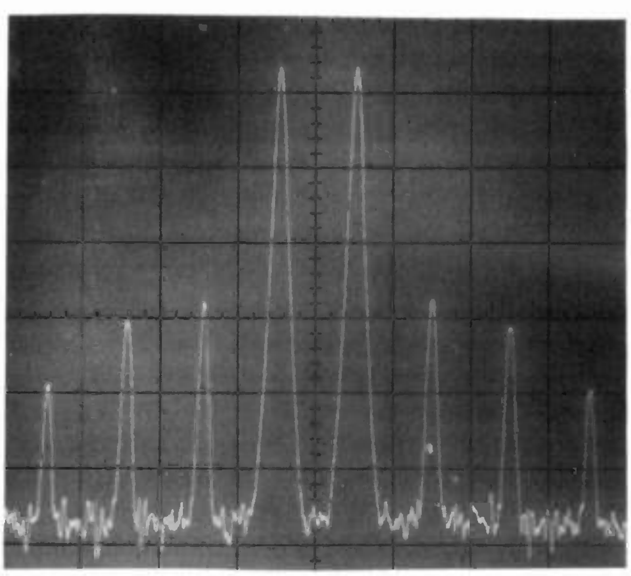


FIG. 15

TH 563 at 31.5 kW -6 dB Δf 0,5 MHz

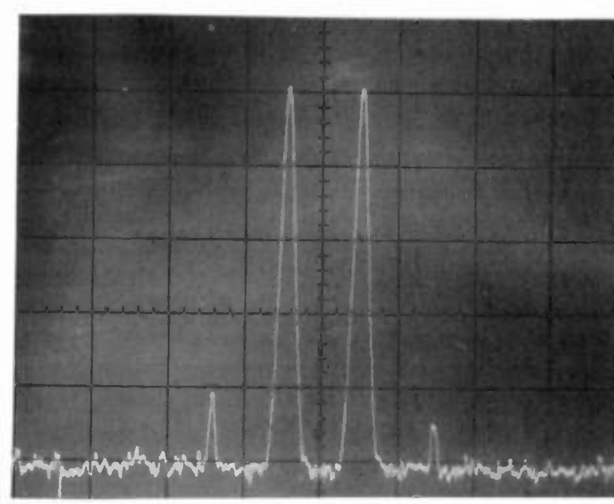


FIG. 17

TH 563 at 31.5 kW -12 dB Δf 2 MHz

IV - TETRODE RELIABILITY DATA

Since many tubes have been in operation at high power level for several years, the high reliability of these tubes has been proven.
In fact, covering 13 countries we can note at the moment :

Nbr	TUBE	kW	AMPLIFIC.
61	TH 582	10	common
174	TH 582	20	visual
4	TH 563	20	common*)
8	TH 563	25	common
1	TH 563	30	common
76	TH 563	25	L standard visual*)

*) under commissioning

These transmitters are built by 9 different manufacturers to the different world standards.
Statistics available from the different end-users after several years give a good idea of tetrode life-time :

for 10 kW common amplific. : 15 000 h
for 20 kW visual amplific. : 10 000 h
for 25 kW common amplific. : >15 000 h

The very good reliability of these tetrodes may be attributed to several parameters :

- Water cooling of the anode and the screen grid connexion which lowers the inner temperature of the tube
- The large safety margin given by the pyrolytic graphite grids for over currents and over heating
- High tetrode efficiency leading to a low dissipated power on the anode
- The stability of the associated cavities which eliminate any variation in power or frequency.

V - CONCLUSION

We have seen throughout this paper that the proven tetrode performances at different power levels are highly consistent with preliminary HDTV specifications. **In particular, the virtual absence of phase distortion and their excellent linearity make these components good candidates for digital transmitters.**
The low level of correction needed, as well as the low anode voltage, simplify design and lower the cost of the transmitter.
The high reliability of tetrodes associated with very simple maintenance (less than 30 minutes to change a tube) minimizes the maintenance cost of the transmitter.
Finally, the good proven lifetime, probably even higher in the case of digital broadcasting, and the low cost of the component make tetrodes the best cost-saving solution for analogical and digital TV broadcasting in this power range.

THE FLEXIBLE USE OF NEW HIGH EFFICIENCY POWER AMPLIFICATION SYSTEMS

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INTRODUCTION

The last four years have seen a proliferation of High-Power amplification devices for use in the UHF TV spectrum (band IV and V). Thompson, Motorola and Philips have given the industry a new generation of High-Power solid-state devices. Varian, Eimac division has initiated the Inductive Output Tube (Klystrode), with EEV following. Varian's microwave tube division has provided the MSDC Klystron, followed by EEV with the ESC, and Philips with the PDC, all using the same basic efficiency improvement technology. If these additions to the marketplace weren't enough, Thompson has expanded the power capacity of its 25kW Tetrode, while we continue to see improvements in the useful life of the tried and true Klystron. All of this is further confused by the availabilities of the Klystrode, IOT and PDC in air-cooled or water-cooled versions.

FACTORS TO BE CONSIDERED

We at TTC have spent significant time in evaluating the optimum use of these devices and feel that each user must evaluate the applications of these technologies for their own use . As one would expect, the following factors of:

- Performance
- Operating costs
- Annual replacement costs
- Capital costs
- Relative maintenance costs
- Operational complexity

are most often considered.

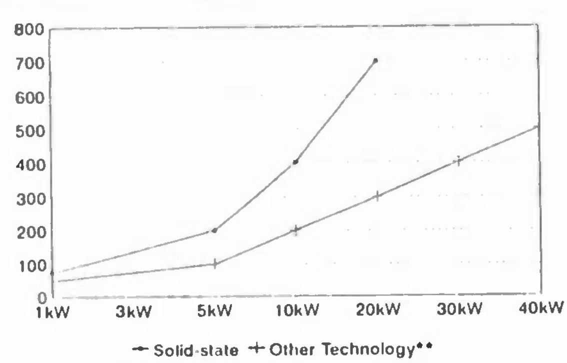
TABLE 1
"New" RF Amplification Devices

AMPLIFIER TYPE	MANUFACTURER	POWER	OUTPUT-COOLING
Solid-State	Philips Thompson Motorola	80-150W	Air/Convection
Tetrode	Thompson	25-30kW	Water/Vapor
Klystrode	Varian	10-40kW 20-60KW	Air Water
ICT	EEV	10-40kW 10-40kW*	Air Water
MSDC	Varian	40-60kW	Pure Water
ESC	EEV	40-60kW 40-70kW	Pure Water Pure Water
PDC	Philips	10-40kW 40-60kW	Air Pure Water

Note: * 60kW under development

While these factors seem as simple as motherhood and apple pie, as we would say in the U.S., our experience has shown a number of subtleties that are not apparent at first glance. For instance, the capital cost of a solid-state transmitter would seem high for the power levels typically used within the European market. However, the capital cost to achieve the same level of redundancy in the other technologies means total system cost is comparable at higher power outputs than first would appear because of the solid-state soft fail characteristic (the ability to lose one or more solid-state output devices without going off-air). Table 2 compares the cost of a solid-state common amplification transmitter and a tube type at comparable redundancy.

TABLE 2
Capital Cost of Solid-State vs. Other Technology*



Notes: * At equal redundancy
** Other technology in combined amplification

PERFORMANCE CHARACTERISTICS

Table 3, shows in simplified format, the nominal performance of several key specifications in both common and separate amplification modes.

As you can see, at a given power level, there is little difference in the differential phase and gain between any of the amplification media. And while some variations occur at the out-of-band performance, these variations are filterable, making the point moot. The two areas that show the most variation are in-band, intermodulation for combined amplification, and cost of operation for all forms.

When you look at the typical In-band Intermodulation (IMD) performance, you see a fairly narrow variation in IMD under the combined amplification mode of the solid-state, Klystron, and the MSDC Klystron. All three of these amplification systems give generally less than optimum performance as combined amplifiers, whereas, the Klystrode, or IOT, and the Tetrode when used in the combined amplification mode typically can reach numbers that are usually specified for separate amplification systems. The effect of Table 3 tends to prove that combined amplification, all else being equal, has certain operating advantages in the region of 20-40kW.

TABLE 3
Key Performance Specifications of Amplification Methods (at 30kW TPO)

Specifications	Solid-State	Tetrooe	Klystrode/IOT	MSDC
Differential Phase (0)	5/3	5/3	5/3	5/3
Differential Gain (%)	2/2	2/2	3/2	3/2
Out of Band Products (dB)	40/60	57/60	40/60	40/60
In Band Intermodulation				
Products (dB)*	40/60	57/60	47/60	47/60
Systems Efficiency (%)**	27/27	28/27	34/49	77/81

Notes: / = Combined Amplification/Separate Amplification
* Numbers quoted are three-tone test and are typically 3dBM improved four-tone
** FOM = peak visual power + aural power/total average mains power

COST OF OPERATIONS

Table 3 also compares the cost of operation, for the five amplification systems. This is expressed as Figure of Merit (FOM). The table reflects very significant variations in the power requirements. The Inductive Output Tube (IOT) has the best FOM for system operation in combined amplification, while the MSDC has a better system operating FOM in the diplexed or separate amplification mode.

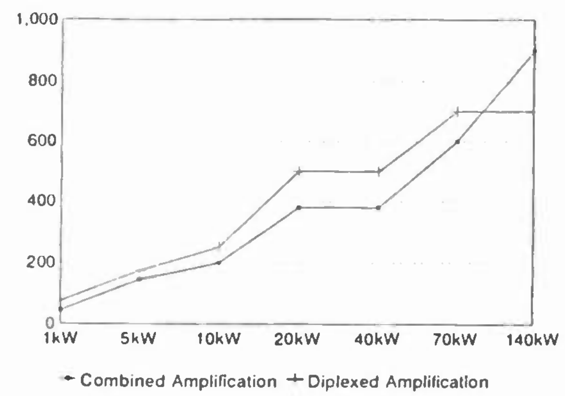
CAPITAL COST

In the capital cost, first glance would indicate combined amplification system to be less costly than separate (diplexed) amplification. If however we review Table 4, which reflects typical capital cost, it shows as you exceed the 70kW range, you begin to reach the crossover point where a diplexed transmitter becomes capital cost advantageous, thereupon leaving you to look again at the comparison in operating costs between an MSDC device and the IOT. The subtleties that now come into play are:

• The current and projected cost of electricity; [The cost per time period = (peak visual + CW aural) ÷ FOM for system x the number of operating hours in the time period x the power rate per kilowatt hour (kwh)]

• The current and projected cost of money. Because of the complexity of defining the cost of capital for purposes of comparison, Table 4 only reflects the raw capital cost and assumes that no matter where it comes from, capital has a comparable cost, whether it is government grant, borrowed or equity.

TABLE 4
Capital Cost of Combined vs. Seperate Amplification



RELATIVE MAINTENANCE COST

There is a prevailing impression that solid-state is simpler and has little cost to maintain. This is an impression we would confirm from our own experience. Because optimization and checks required for high-power water-cooled Tetrodes, the maintenance factor is perceived high. Again, because of the high voltage involved and the prevalence of water cooling with conventional Klystrons, they too would be classified in the same area as the Tetrode. If you use the air-cooled version of the Klystrode or IOT, our experience would indicate a slightly lower maintenance involvement than a water-cooled Klystrode IOT. But it still remains of some concern since high voltage is involved and the system must necessarily, in the case of the Klystrode, be concerned with the process referred to as "scrubbing" (a process that is disappearing as the product matures). Because of the purity of water requirement of the MSDC, and the inherent concerns over cooling the multiple depressed collectors while maintaining the separation of those voltages, we place the MSDC on the high end of the maintenance scale.

TABLE 5
Relative Maintenance Requirement

COMPLEXITY	AMPLIFIER-TYPE
10	MSDC, water-cooled
7	Klystron, Klystrode /IOT, Tetrode, water-cooled
5	Klystrode /IOT, MSDC, air-cooled
3	Tetrode, air-cooled
2	Solid-State, air-cooled
1	Solid-State, convection-cooled

Notes: 10 = Most Complex/1 = Least Complex

ANNUAL REPLACEMENT COST

Table 6 shows the normal annual replacement cost for each of these modes of amplification. As you examine this Table, keep in mind that while there is a perception of no replacement cost in solid-state, this is not true. It is our experience that when you magnify the mean time to failure rate by the number used in any one unit, that there is a fairly linear progression to the failure rate; this failure rate is not always the result of failures of the device, but can be as a result of failures in other portions of the system. The advantage of these failures is their occurrence in the soft-fail mode, i.e., only the failure of one device or module. The disadvantage in solid-state is the normal requirement in the event of failure to change more than one component/device in the module. Here we have attempted to forecast, based on industry data, the annual replacement cost for solid-state and other amplifier systems. We have assumed a 24-hour operating day and utilized the vendor's U.S. published list prices to develop the data.

OPERATING COMPLEXITY

It is our experience that operation complexity is in the same order as maintenance cost, with the most complex product being the MSDC, with the second most complex being the water-cooled products. The air-cooled products are less complex to maintain than water, while the solid-state products have the lowest maintenance complexity.

TABLE 6
Projected Annual Replacement Cost

Power Output	Solide State	Tetrode	Klystron	IOT	MSDC
1kW	0.6	4.7	N/A	N/A	N/A
5kW	1.2	10.5	N/A	N/A	N/A
10kW	2.4	11.1	4.5	9.4	9.6
20kW	4.2	15.2	4.6	9.4	10.4
30kW	6.2	15.2	6.9	9.4	10.4

Device(s), U.S. Dollars in Thousands.

CONCLUSIONS

This leads us to the following general conclusions related to the evaluation factors.

• At power levels of one to five kilowatts, the competition is between solid-state and Tetrodes. Tetrodes have a lower capital cost, but both products have nearly equal operating costs. Tetrodes, on the other hand, have a significantly higher hourly replacement cost than solid-state devices. Therefore, over the long term we can conclude that solid-state is a major advantage where the user can meet the capital cost.

• In the power level ranges between five and ten kW, the decision gets less clear, but the same principles do not apply. The capital cost becomes fare more dominant and the operating cost (power and amplifier life) are the more variable parts of the equation. Examining these, there is still significant merit to the use of solid-state, but Tetrodes become serious contenders. At 10kW and above, the situation is further complicated by the addition of the air-cooled Klystron, air-cooled MSDC, and the air-cooled IOT. Initial cost considerations for these three products are significantly lower than the cost for solid-state, although marginally higher than the cost of for the Tetrode. Performance of the Klystrode /IOT is comparable to the Tetrode, particularly in combined amplification with its reduced capital and operating cost.

• Also, as we increase output power above 10kW, there is a crossover between power cost and capital cost. It is clear that somewhere between five and ten kW, the high capital cost of solid-state coupled with this relative inefficiency, makes it less and less of a contender. As you increase the cost of power, the greater the impact of the power cost to the equation and the more we are driven towards the Klystrode /IOT or MSDC technology.

• Above 90kW, because of the capital and operating cost factors and in spite of the maintenance and complexity considerations, the MSDC technology begins to take over.

• In the large range between 10 and 90kW, the use of the Klystrode/IOT a very serious contender.

USING THE INFORMATION

How do you put this all to use? First you must remember, in developing your comparison of various output devices, to get the same order of redundancy as solid-state with discrete devices such as an IOT, the discrete device must be used in parallel.

With this in mind, in a recent TTC project, we looked for a method of solving the following challenges:

- Redundancy
- Lowest operating cost
- Common output device at 10-20 and 30kW TPO

- Lowest or comparable capital cost
- Guarantees of fully competitive performance

Utilizing the flexibility of all the amplifier systems we've just described, we decided to use a visual IOT diplexed with a solid-state aural as the primary transmitter. A secondary IOT amplifier was provided as a redundant visual amplifier. To further enhance the system, the redundant visual amplifier was made available full time in common mode amplification to replace the entire diplexed system if necessary. The final result of this mixed use of solid-state and tube amplification was to achieve at the required levels, competitive capital cost with redundancy and relative low operating cost. In fact, the system has been so successful, the customer may remove the solid-state aural units and convert them to transposers as the performance of the IOT at combined amplification is so good and the cost of operation is so low. In another application, we were able to diplex a Tetrode with a solid-state aural to achieve the same set of goals.

Obviously, there are many variations that can be achieved in use of all these technologies. As you have seen, each have their advantages and disadvantages. In our view, the new technologies now offer us a wide range of options to meet the individual needs of the Broadcast Community.

PRACTICAL IMPLEMENTATION OF TRANSMISSION LINES AND ANTENNAS FOR THE SIMULCAST PERIOD FOR NTSC/HDTV

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ABSTRACT

For the transition from NTSC to HDTV, multi-user tower situations will provide the greatest challenges. This paper is intended to present a method for beginning the discussions regarding simulcast transmissions with co-users. Based on existing transmission line and antenna designs, the following factors are discussed: tower structural loading, antenna types, multichannel vs. single channel designs and coaxial line vs. waveguide. More importantly, the interactions between these factors are described so that engineering staffs can do preliminary design "what ifs".

INTRODUCTION

Perhaps the most serious logistical problem involved with the proposed Simulcast Period will be finding a location to mount an additional antenna and transmission line needed for the HDTV channel. Finding space on towers where a single TV broadcast antenna is mounted will typically be a matter of determining what additional loading the tower can withstand and making room for the new antenna. On the otherhand, towers where multiple broadcast antennas are located will involve not only structural engineering concerns, but will require a cooperation between the users that will rely heavily on the electrical performance options available for the antennas and transmission lines. The following information is intended to provide the groundwork for more detailed discussions during the planning stages for a Simulcast Period.

Tower Concerns

The first issue that should be addressed is the ability of the tower to support additional loading. This involves both the magnitude of the additional load and its location on the tower. Typically, the lower on the tower the additional load is placed, the easier it is for the tower to accept the load. Using typical antenna windload values and transmission line sizes (See Table 1), several configurations can be determined from the results of a tower structural analysis. Obviously, the coverage requirements for the viewing area will play an important role in deciding which configuration is optimum. It should also be noted that changes in the structural codes since the original installation of the tower may have dramatic effects on the allowable loads and their placements. Using a qualified structural consultant is a must to avoid problems later on.

Antennas and Coverage

(Note: For this discussion, it is assumed that UHF frequencies are used for the HDTV channel.) Once the locations of additional loading to the tower are determined, a more difficult task now must be addressed: A choice of antenna radiation patterns must be made that will best serve the coverage requirements of the individual stations. Since this discussion is primarily focused on multi-user towers, it is assumed that most of the stations will be sidemounted to the tower using directional antennas (either by design or as a result of tower interference). In these cases, a directional antenna for HDTV would seem to be a logical choice. A directional pattern requirement will allow more latitude in the placement of the antenna, including the possibility of two or more antennas at the same level of the tower (See Figure 1).

In the case of a station presently using a top mounted, omnidirectional antenna, the choice of antenna pattern is not as clear. However, it should be remembered that in the beginning years, the penetration of HDTV receivers will be relatively small. In many cases, the viewing population where these receivers will originally emerge can be determined and a directional pattern will suffice. As the number of HDTV receivers increases, there may be a point in time when replacing the top mounted antenna with an HDTV channel antenna can be justified. In this case, the NTSC channel would then be transmitted from a directional sidemounted antenna. While this scenario may appear to produce additional costs over the discussion above, the replacement decision will be based primarily on the stations's coverage needs. Until the Field Tests are performed on an ATV system, the final requirements for long term transmission of HDTV will not be known. Low cost and low windload NTSC and HDTV compatible sidemounted antennas are presently available[1] and may be applicable for both interim and long term use as the HDTV antenna.

MultiChannel Designs

Several discussions[(2),(3)] have previously occurred regarding multichannel antenna designs for transmitting HDTV. The primary benefit is a reduction in the number of transmission lines required in the tower. It has been shown that in many cases, particularly taller towers, the transmission lines present a substantially greater load to the tower than the antennas. Therefore, reducing the number of transmission lines required may be the only way to allow all users on a tower to simulcast from that tower. The primary restrictions to the use of a multichannel antenna are that each stations's coverage requirement must be the same and the number of stations per antenna will be based on the maximum power rating of the antenna or transmission line.

Transmission Lines

The transmission line requirements will be based on the final transmitter power levels. In most cases, the average power will be the determining factor in the choice of transmission line. Because it is anticipated that the average power level of a digital HDTV signal will be from 6dB to 10dB below the present NTSC levels, smaller transmission lines can be used. And since the required ERP levels will be lower due to the lower signal/noise requirements, the additional attenuation of the smaller lines will be acceptable.

Regarding the use of waveguide, the additional windload generated will likely eliminate this option from most installations, except for multi-channel antennas. For this case, the channels used must be chosen carefully for optimum performance.

TABLE 1
Typical Transmission Line Sizes
and Antenna Windloads

Tx Line Type	Outside Dia. (inches)	Attenuation (dB/100') @ 600 MHz	Average Power (Kw)
Air Semiflexible:			
1 5/8	1.98	0.528	5.62
2 1/4	2.38	0.442	7.66
3	3.02	0.409	13.0
4	4.00	0.320	20.3
5	5.20	0.212	27.5

Ant. Gain (RMS)	Outside Dia. (inches)	Windload (lb)@50/33	Average Power (Kw) @ 600 MHz
16 (12.04db)	3.5	740	16
24 (13.80dB)	3.5	1100	22
32 (15.05dB)	3.5	1470	22
25 (13.98dB)	12	2700	60
30 (14.77dB)	12	3100	60

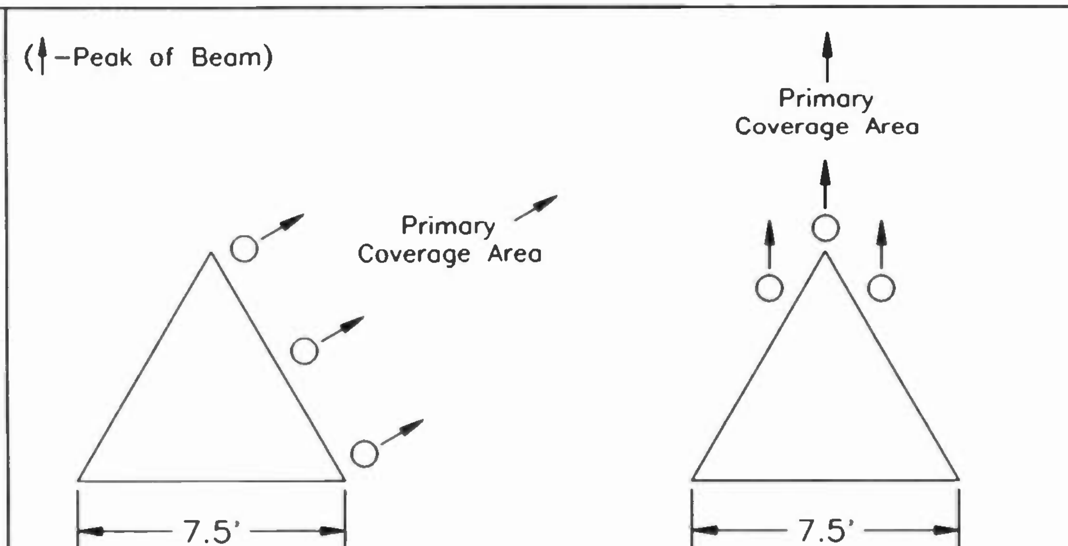

Figure 1
Multiple Directional Antenna Configurations for Efficient Tower Space use

System Integration

Based on the previous discussions, it is apparent that a multitude of options are available for implementation of an HDTV channel. In the case of single channel implementation, it will be necessary to use the smallest transmission line size available for locations high on the tower. This will require a high gain antenna to reach a specified ERP. As the antenna is located lower on the tower, larger line can be used allowing higher power and less attenuation and a lower gain antenna. It should also be noted that the lower ERP anticipated for coverage with a digital signal will allow locating the antenna lower on the tower when the prime coverage area is not limited by the radio horizon.

For multichannel implementation, larger transmission lines will be required for power handling with the possible result of locating the antenna lower on the tower.

SUMMARY

Several ideas have been presented regarding the implementation of HDTV transmission on existing multi user towers. The intent of the scenarios described was to present alternative concepts to the "standard" philosophies that have been used for over 30 years. For a successful implementation of a new transmission system to occur, discussion and cooperation between the station marketing and engineering functions with the product suppliers are imperative. New concepts based on an understanding of all these functions can result in more cost effective methods making the change more financially available to all broadcasters.

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DUAL USE™ UHF TRANSMITTERS FOR THE HDTV SIMULCAST PERIOD

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This paper will discuss the transmitter requirements for the HDTV simulcast period. The paper will describe the features of certain transmitter types and configurations that permit future reconfiguration into two independent transmitters. One transmitter for NTSC and the other for D-HDTV without significant loss of rated output power or signal quality.

These **DUAL USE™** transmitters will be described in detail using examples of actual installed hardware.

I. INTRODUCTION

The U.S. broadcaster is faced with a challenging period during which he will be required to transmit both today's NTSC signal and a new digital HDTV (D-HDTV) standard based on video compression technology. The current plan calls for transmissions to take place simultaneously for a period of ten years, beginning approximately five years from now.

There are many equipment decisions that will face the broadcasters. One of the first will be what kind of transmitter will be required and how can the purchaser avoid early equipment obsolescence. The transmitter may be the first piece of equipment to be purchased since the future ramp up of the consumer receiver population requires the presence of an on-the-air signal. Many D-HDTV program sources will be satellite delivered ready for transmission by simple "pass through"[1] facilities.

Two other factors driving the equipment purchase decision are:

1. All NTSC transmission will be scheduled for shut down by 2008.

2. The vast majority of D-HDTV channels will be UHF.

These factors are particularly important to broadcasters who must replace aging UHF equipment today and VHF broadcasters viewing a UHF future.

Since D-HDTV will be primarily a UHF service, a practical hardware solution in UHF will now be examined.

II. NTSC COMPARED TO D-HDTV

The ideal solution for today's UHF broadcaster using aging transmitting equipment needing replacement, is a system that is capable of practical operation as either an NTSC or D-HDTV transmitter. We, at Comark, have named this type of service **DUAL USE™**.

In order to identify the ideal **DUAL USE™** system, it is necessary to examine the general characteristics of both NTSC and D-HDTV signals.

The differences between the NTSC signal and the D-HDTV signal are significant. The NTSC signal has a high peak power that occurs predictably. Further, the NTSC signal has a peak to average power ratio that varies with picture content but can be as low as 2.3db.

These characteristics place a heavy average power burden on the NTSC transmitter output stage and supporting circuitry due to the high duty cycle of the signal.

The D-HDTV signal is essentially a broadband noise signal[2] with a constant average power and a large peak to average ratio. In other words, a low duty cycle. Further, the peak power of the D-HDTV signal occurs randomly in time and frequency. The D-HDTV signal is easily damaged by non-linear distortions in the power amplifier stages of a transmitter. Distortions that create inter-modulation products are particularly damaging. Such distortions in the high power amplifier of a D-HDTV transmitter will reduce the transmitter's effective range[3], while in an NTSC transmitter, the distortion only affects picture quality.

Thus, a transmitter for today's NTSC signal must handle a high average power signal compared to the D-HDTV transmitter which must handle a lower average power but be capable of linear amplification of short randomly occurring peaks of perhaps ten times the average output power.

Figure 1 is a spectrum of the ATRC AD-HDTV signal broadcast by a Comark transmitter in September 1992 from the WRC-TV facility in Washington, D.C. It is generally representative of the type of signal the D-HDTV transmitter will be required to handle. Depending on the final D-HDTV system chosen by the FCC, the peak power required for coverage may be equal to or up to 6db less than today's NTSC levels.

III. DUAL USE™

Given the different natures of the NTSC and D-HDTV signals, is it possible to build a transmitter that can provide uncompromised performance in both formats? The answer is yes and this paper will explore that answer.

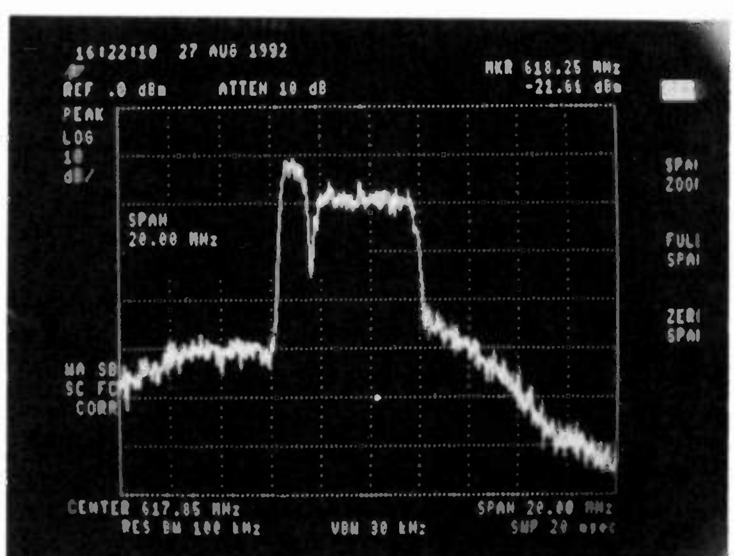

Figure 1 - Spectrum of the ATRC AD-HDTV Signal

Historically, the high power UHF transmitter has been equipped with some version of a klystron. Several papers[2,4,5] have already explored the klystron and its derivative, the MSDC or ESC klystron, as candidates for D-HDTV amplifiers. The conclusions for the klystron are that it may technically provide D-HDTV service, but at a high price in energy consumption, size, complexity and cost.

The Class A nature of the klystron makes it very energy inefficient when amplifying a signal with a low duty cycle like D-HDTV. It has been shown that the best efficiency in D-HDTV will be achieved when a Class AB device is used as the power amplifier[4,5]. The Class AB device will draw power as a function of the average signal level and thus, its power consumption will reflect the low average power of the D-HDTV signal.

Such Class AB devices are tetrodes, transistors and inductive output tubes.

The solution for today's NTSC requirements of 60 kW and higher rules out the tetrode and transistors for dual use applications.

The inductive output tube represents a family of devices that include the IOT and the Klystrode® with individual single tube ratings of up to 60 kW.

In order that a transmitter be truly capable of **DUAL USE™**, the RF output system should be free of bandwidth limiting components like notch diplexers. Fortunately, the inductive output tube has proven to be very effective as a high quality common amplification device[6], amplifying both the vision and sound carriers simultaneously in a single output tube. This application of the inductive output tube eliminates the output diplexer and creates the possibility of a true **DUAL USE™** system.

IV. THE DUAL USE™ SOLUTION

Figure 2 is a block diagram of a **DUAL USE™** configuration. The intent of the system is to permit full specification operation as an NTSC transmitter with the future possibility of conversion to D-HDTV with minor modifications. If, as is shown in Figure 2, a parallel system is chosen for NTSC operation, it is possible to split the system apart into an NTSC and D-HDTV transmitter. Thus, two separate systems on different channels can be created for the simulcast period.

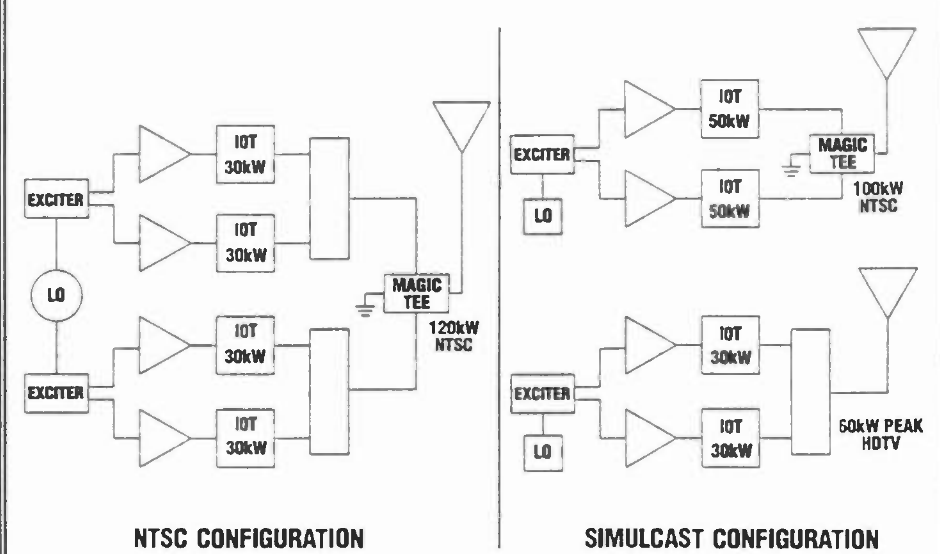

Figure 2 - Block Diagram of a DUAL USE™ Configuration

V. THE DUAL USE™ TRANSMITTER IN NTSC

NTSC operation of a true **DUAL USE™** transmitter will utilize common amplification technology. Since all power amplifiers introduce some distortion if they are used efficiently, intermodulation can be expected between the visual and aural carriers in common amplification.

The technology has existed for many years to precorrect the input signal to a power amplifier in order to eliminate time domain distortions. Such precorrection also reduces intermodulation to some degree.

If true broadcast quality signals are to be broadcast in common amplification, it is necessary to achieve intermodulation suppression of at least -58db below the peak sync power. Such a level of correction requires that the issue of broadband phase linearity, as well as amplitude linearity, be addressed. Careful attention must also be exercised to avoid designing cascaded Class AB amplifiers[2] into the system.

Two Class AB stages in cascade create a complex small signal turn on curve that is difficult to correct, in a stable fashion, to -58db intermodulation levels. While correction may be possible at a single operating point, a minor change in transmitter operating parameters will require readjustment of the linear corrector. This is undesirable for broadcast service.

Comark has developed a linearity correction system that maintains the IM correction across the entire channel bandwidth. This broadband, non-frequency selective (N.F.S.) corrector provides IM correction to beyond -62db on a long term stable basis.

An example of this technology is a Comark IOX™ transmitter equipped with the corrector that was recently delivered to the Finnish Broadcast Company for service in two carrier NICAM sound in common amplification. This system utilizes two sound carrier plus the visual carrier. -60db or better IM levels were achieved over the required PAL bandwidth. Such performance could not be possible with a standard linearity correction system.

Figure 3 is a spectrum photo of the fully corrected dual carrier NICAM common amplification output at 25 kW peak sync from the Comark IOX transmitter. The linearity performance necessary to meet -60db IM levels is fully 10db better than that necessary to preserve video waveforms. Thus, if the IM levels are achieved, the video signals from the transmitter are nearly textbook perfect. Previous papers have demonstrated the signal purity of a fully corrected common amplification transmitter[6].

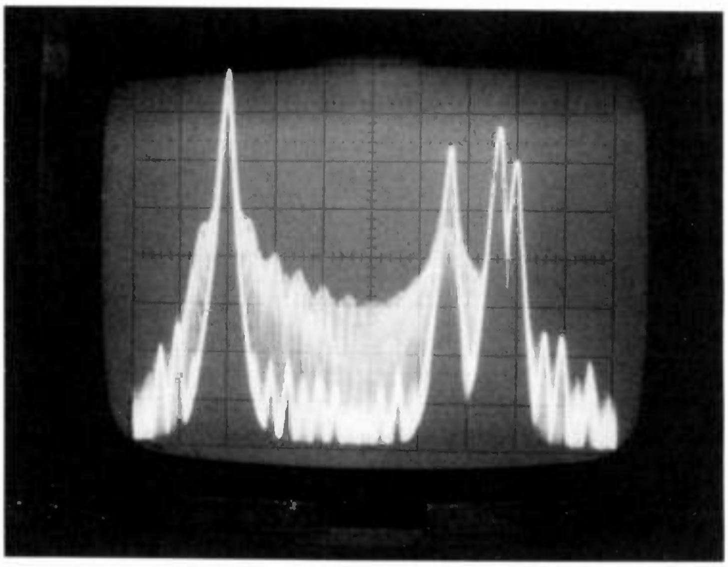

Figure 3 - Fully corrected dual carrier NICAM common amplification output at 25 kW peak sync from the Comark IOX™ Transmitter

Common amplification for broadcast service also requires that the aural carrier be corrected for cross-modulation from the horizontal sync of the visual signal[2]. The H sync signal on the visual carrier drives the power amplifier to its maximum output power. Energy efficiency considerations and economics require that at maximum output power, the amplifier is not perfectly linear. While the visual corrector will restore the sync signal, the aural carrier is also being distorted in both amplitude and phase.

The aural carrier distortion in the phase domain creates unwanted equivalent peak deviation of the aural carrier at 15,734 HZ and its higher order harmonics. This cross modulation will violate two FCC rules if not corrected. One rule deals with protection of the stereo pilot frequency and the other deals with overmodulation of the aural carrier. (See Figure 5)

While FCC enforcement of the stereo pilot protection rule has been less than aggressive, the overmodulation of the aural carrier will attract the FCC's attention. Spurious modulation of as much as 50 KHZ has been measured from just the H sync cross-modulation with no audio input to the transmitter. Aural cross-modulation has the potential to eliminate common amplification as a viable transmitter option for full broadcast service.

Fortunately, it is possible to correct the aural carrier in both amplitude and phase to eliminate any problem with either over-modulation or stereo pilot protection. Comark holds a U.S. patent on the system that is necessary to achieve this correction. A description of the system and its results are covered elsewhere[2].

Therefore, in NTSC service, a **DUAL USE™** transmitter in common amplification will provide superior signal quality for both the visual and aural carriers. The efficiency of the **DUAL USE™** system defines the present state-of-the-art. The Class AB nature of the inductive output tube permits the transmitter to perform at full power at plant efficiencies of 80% or better. There is no sacrifice of energy efficiency to obtain **DUAL USE™** capability and superior signal quality.

VI. THE DUAL USE TRANSMITTER IN D-HDTV

The low duty cycle of the D-HDTV signal and its requirement for low intermodulation amplification is ideal for the Class AB linearity corrected inductive output tube transmitter. It is clear that the Class AB nature of the inductive output tube technology will allow the transmitter to take advantage of the signal's low duty cycle and be extremely efficient.

A broadband linearity correction system, like that in the Comark **DUAL USE™** system, is required to insure that a low level of inter-modulation distortion is maintained at any frequency within the D-HDTV bandwidth.

As a result of the close channel spacing, the D-HDTV service will require excellent out-of-band energy reduction. The transmitter's linearity correction system does reduce out-of-band IM products. However, additional bandpass filtering is required. Simple notch filtering is not applicable.

If the output circuit of the inductive output tube is connected to a normal bandpass filter assembly, serious problems can be created. The impedance of the bandpass filter will not be constant over the much broader bandwidth of the transmitter's output stage. Thus, serious mismatches resulting in bandpass anomalies in amplitude and phase will be created. Such anomalies are not acceptable to the D-HDTV signal.

Therefore, if bandpass filtering is to be required for D-HDTV, then a constant impedance, over a wide bandwidth, output filter is required.

Figure 4 is a block diagram of Comark's constant impedance, balanced output filter. This filter provides harmonic rejection and out-of-band IM rejection in NTSC and broadband energy rejection in D-HDTV, while also presenting the output amplifier of the transmitter with a broadband constant impedance.

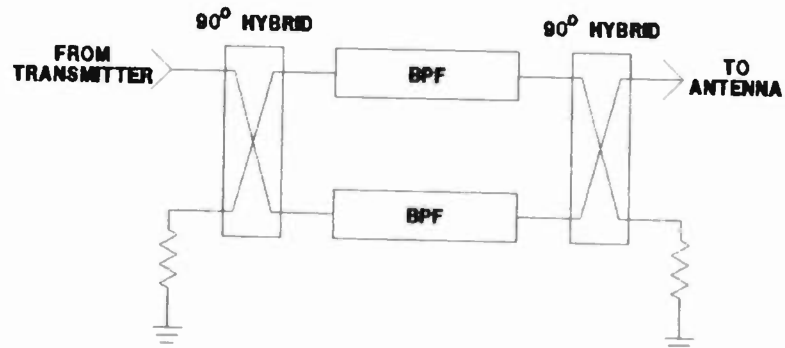

Figure 4 - Constant Impedance Output Bandpass Filter

IOT DUAL USE™ TRANSMITTER

NTSC SPECIFICATIONS	
Output Power	50 kW Visual, 2.5 kW Aural
Frequency Range	470-806 MHz
Driver Amplifier	Solid-State Class A
Low Frequency Non-Linearity	0.5db
Differential Phase	1°
Differential Gain	0.5db
Common Amplification Applications In Band Intermodulation	-60db
Stereo Pilot Carrier Protection and FM Overmodulation	Full compliance with FCC Specifications 73.682(c)(3) and 73.157(b)(3)
AD-HDTV SPECIFICATIONS	
Power Output (Peak/Average)	50 kW Peak, 5 kW Average (Calorimetric)
Bit Error Rate (BER) (50 kW Peak/5 kW Avg.)	Better than 1 x 10^{-10}
Intermodulation Performance (Peak/Average Out-of-Band)	Better than 60db
DC/RF Conversion Efficiency (Average RF Power)	25%
Amplitude Linearity over a 30db Dynamic Range	$\pm$0.2db max. $\pm$0.1db typ.
Phase Linearity	$\pm$1.0° max., $\pm$0.5° typ.
Group Delay	$\pm$25nS max., $\pm$10nS typ.
Cooling Requirements	5 GPM @ 5kW Avg. power
Dimensions	102" L x 30" D x 70" H
Output Tube	EEV 60 kW IOT
Output Filter	Constant Impedance Bandpass Type

Figure 5 - Performance Characteristics of Comark DUAL USE™ Transmitter System

Figure 5 is a chart listing the performance characteristics of the Comark **DUAL USE™** transmitter system in both D-HDTV and NTSC service.

VII. FIELD OPERATION

Comark has installed over eighteen transmitters using thirty-seven IOT tubes capable of **DUAL USE™**. These transmitters have accumulated over 300,000 tube hours in NTSC and have proven the performance of the system in terms of signal fidelity, reliability and efficiency.

In September of 1992, Comark supplied an "S" series **DUAL USE™** transmitter to support the ATRC tests at WRC-TV in Washington, D.C. The transmitter provided 5 kW average, 50 kW peak AD-HDTV signal, on Channel 38, and with a switch in the IF input, provided 50 kW peak sync NTSC output. This was done without any adjustment to the transmitter between NTSC and AD-HDTV. The test results at WRC-TV are covered in another paper[7]. However, the Comark **DUAL USE™** system proved itself by providing superior NTSC pictures and also establishing a 70 mile range benchmark for AD-HDTV using a consumer receiving antenna at 30 feet above the ground. This range was achieved in large measure because of the low intermodulation distortion from the transmitter.

Figure 6 is a view of the WRC installation showing the single ended Comark **DUAL USE™** "S" series transmitter. The linearity correction system and low level components are located on the left side of the transmitter. This transmitter has since been shipped to Charlotte, NC to serve as the UHF standard test transmitter during the Advisory Committee on Advanced Television Service (ACATS) field tests.

Figure 6 - Comark DUAL USE™ "S" Series Transmitter - WRC-TV

VIII. NEXT GENERATION DUAL USE™ - THE IOX

In a continuing effort to take advantage of the new technology, Comark has developed the next generation inductive output tube transmitter. Named the IOX™ line, the transmitter features broadband linearity correction of both the visual and aural carriers, full compliance with international safety standards (IEC 215), internal a.c. power distribution to relieve installation complexity, positive pressure cabinet cooling to reduce maintenance, opto-isolated CMOS control logic and full **DUAL USE™** capability including a constant impedance output bandpass filter. The IOX line also can be supplied with internal air dielectric or external unitized high voltage power supply configurations.

Figure 7 is a view of a single ended IOX transmitter, with its internal HVPS, rated for 50 kW NTSC and 5 kW average AD-HDTV. This building block can be configured to be combined to produce transmitters rated for up to 200 kW in NTSC service and 20 kW average in D-HDTV*.

Figure 8 lists the important features of the IOX transmitter line.

More information on the new product line is available directly from Comark.

* Note: The actual average power rating in D-HDTV will depend on the final system chosen by the FCC.

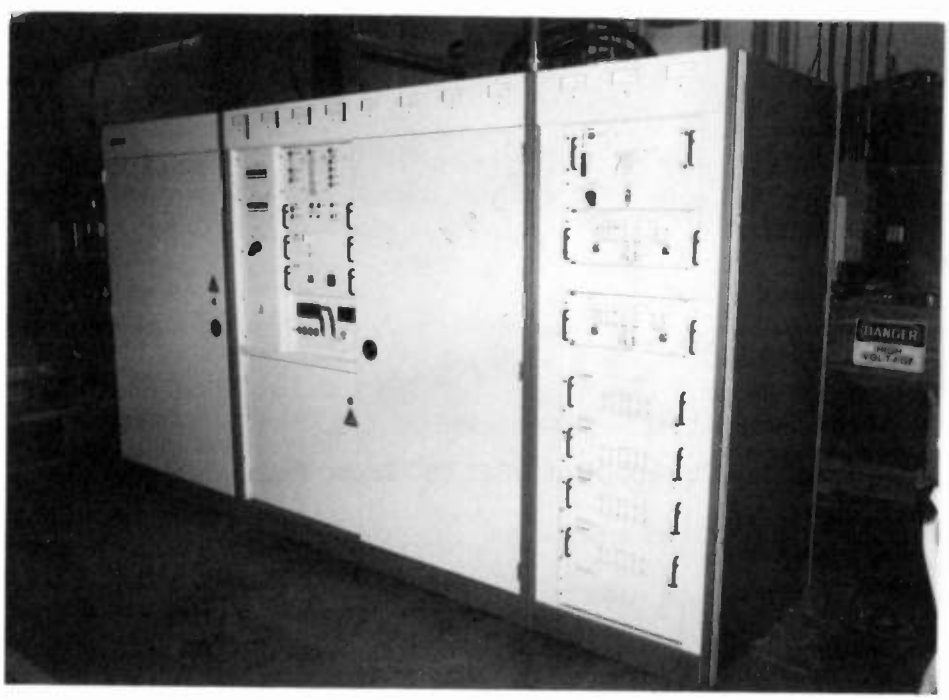

Figure 7 - Single Ended IOX™ Transmitter

IOX™ Transmitter Line
Features

- Full **DUAL USE™** capability
- Power levels from 20 kW to 200 kW
- Broadband linearity correction on both visual and aural carriers - meets all FCC specifications for aural deviation modulation purity
- Compliant to IEC 215 International Safety Design Standard
- Optically isolated C-Mos control logic
- Internal a.c. power distribution and SCR switching
- Alternative H.V.P.S. configurations
- Advanced crowbar
- Solid-state Class A drivers
- Positive pressure cabinet cooling
- Constant impedance output bandpass filtering

Figure 8 - IOX™ Features

IX. CONCLUSIONS

The coming D-HDTV revolution creates the need for a transmitter system that can provide optimum performance in today's NTSC world and tomorrow's D-HDTV with a minimum of modification or additional cost input. This paper has shown that such a system is technically possible. With careful design, it is possible to satisfy the most demanding NTSC requirement today and be ready for D-HDTV tomorrow.

DUAL USE™ systems are already in service and the next generation of equipment has just been introduced.

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RESPONSE OF THE IOT HIGH POWER AMPLIFIER TO THE DIGITAL FORMAT HDTV SIGNAL UNDER FULL POWER TEST

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This paper reviews Television Technology Corporation contributions to the HDTV Field Test Committee, General Instruments and to Zenith. We encompass the Milwaukee to Chicago Zenith test, the lab test on the Zenith system using a 40 KW IOT amplifier at full power showing the efficiency and performance and an explanation on what can be done to be HDTV compatible transmitters now.

The HDTV Committee and the FCC decided to leap frog the Japanese and European approach in high definition television. Current FCC plans for HDTV call for simulcasting the all digital HDTV signal on a UHF channel. Channel assignments have been worked out by the Rule Making body allotting frequencies to each market and allowing the stations in each market to work out the actual assignment of the channels.

There are four digital HDTV encoding systems still in competition for the new standard. One is from Zenith, another are two from General Instruments/MIT, a third is from the North American Operation of Philips and the fourth is from the Sarnoff Research Center. All have ghost canceling and data compression schemes which appear to be robust. The Zenith, GI and Philips use the AT&T digital compression system. TTC has worked with the HDTV Field test committee and Zenith to provide data on high power television's capabilities. At the last NAB, the General Instruments DigiCipher system was publicly broadcast for the first time through a TTC XL-20MU solid state transmitter.

The FCC originally had proposed specifications for transmitters which were extremely tight and which required better equipment than was available four years ago. However, the recent development of the IOT/Klystrode tube has enabled high power transmitters to be more efficient and linear even though they are class "AB" amplifiers. By definition, in the rest of this paper IOT/Klystrode will be stated as IOT for its ease in spelling.

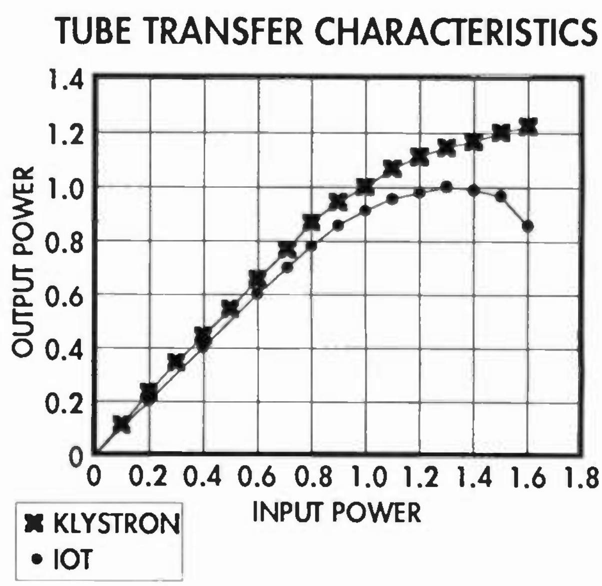

Figure 1 - Transfer curve

The IOT ability to pass the HDTV signal with the required high peak power requirement is made possible by the gradual knee on the tube's transfer curve as shown in figure 1, so that peak power out is limited by pre-correction, grid dissipation and the RF high voltage flash over point. State technology has also made strides with the improvement in pre-correction capability.

Common carrier transmission has become practical as pre-correction can make the transmission linear and bring both in and out of band intermod products to an acceptable level, -54 dB and often exceeding -62 dB.

There are considerations which HDTV systems impose on the transmitter. The HDTV signal has the peak power variations which vary from 8.5 dB to 10.1 dB higher than average depending on the sequence. Dr. Yiyan Wu of COC of Canada, Research Center reports the following peak to average ratios. GI DigiCipher - 9.3 dB, ATRC ADTV -9.6 dB, MIT/GI - 9.9 dB, and Zenith - 10.1 dB. Zenith digital correction is done with a sample of the output of the transmitter and fed through a sample HDTV receiver into a resident computer, a personal type computer, which is external to the normal transmitting equipment. The computer feeds a change to the digital modulator which modifies the signal accordingly. The digital modulator system is then returned to normal as this correction occurs only once. The bit errors which occur on the digital signal do not appear as snow or color distortions but may appear somewhat like a kaleidoscope presentation.

To simulate the HDTV signal for analyzing both in and out of band products, we use a Two-tone test with the second tone being 3 MHz away from the first and then FM deviated by 3 MHz. Intermod of both in and out of band products will then equal the digital signal. Corrections done under these circumstances may achieve the best results for HDTV testing. Figure 2 shows the spectrum of the IOT transmitter's output using the Zenith system uncorrected.

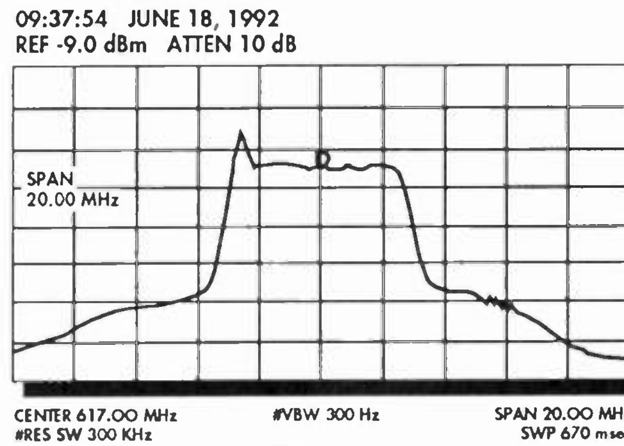

Figure 2

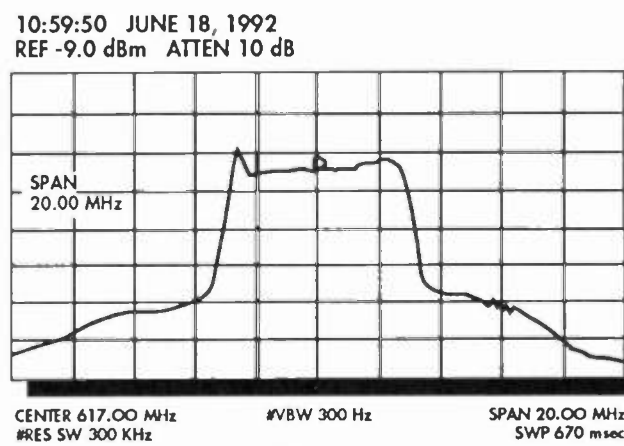

Figure 3

Figure 3 shows the same system digitally corrected. The tilt seen here was necessary to correct for the receiver in this case. The out of band products shown are -33 dB down, unfiltered or pre-corrected, from the average power level and -42 dB down from actual peak power output. With an MCI band pass filter, the out of band products would be reduced 33 dB 2MHz from band edge. As you can see, there is a pilot/sync area which is at the peak.

TTC participated in the recent over-the-air test where the Zenith digital HDTV signal was transmitted from Milwaukee to Chicago. The test was performed on a standard klystron transmitter using a TTC IF Pre-corrector and Up-converter with a 6 Watt output. The output was fed to a pair of 30 KW Klystron amplifiers combined to run at an average power of 10 KW which translates to between 47 KW and 70 KW peak power depending on the digital sequence and Klystron compression. Each tube was running with a beam voltage of 23 KV and current of 3 Amps or 69 KVA. If we were to use MSDC tubes , the power input would be reduced from 138 KVA to 80 KVA. This signal was transmitted to Chicago, approximately 75 miles away, where there was also a Low Power station transmitting NTSC on the same channel. Before the test in Chicago, the reception of the analog NTSC signal from the test transmitter in Milwaukee resulted in a very noisy picture even with the transmitter's power output being at 60 KW peak level and the co-channel signal off. The curve "B" contour was at 45 miles. With the co-channel on, the reception showed the usual co-channel artifacts. But, even when the digital HDTV signal power was varied from 10 kw to below 700 Watts average power, reception was clear and no errors occurred. The potentially interfering Low Power NTSC transmitter was turned on and off and no change in the results was noted.

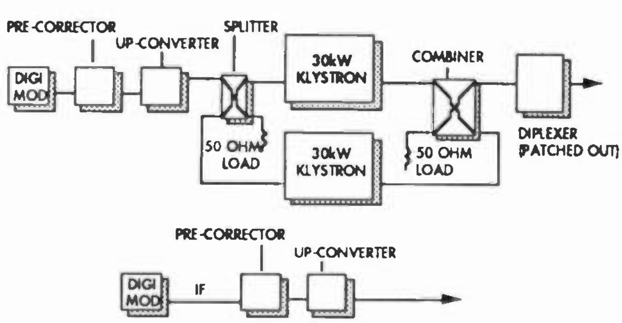

Figure 4

Figure 4 shows the transmitter blocks used in Milwaukee and the system used at NAB 1992 with General Instruments.

We submitted IOT Test Data to the HDTV Committee. The test data presented the results of using the same type of IF pre-corrector and Up-converter as used in the Milwaukee to Chicago test. This was the input for this test of a TTC UHF-30MA, IOT transmitter operating at 30 KW peak power. The data confirmed the Group Response characteristics from 1.5 KW to 30 KW.

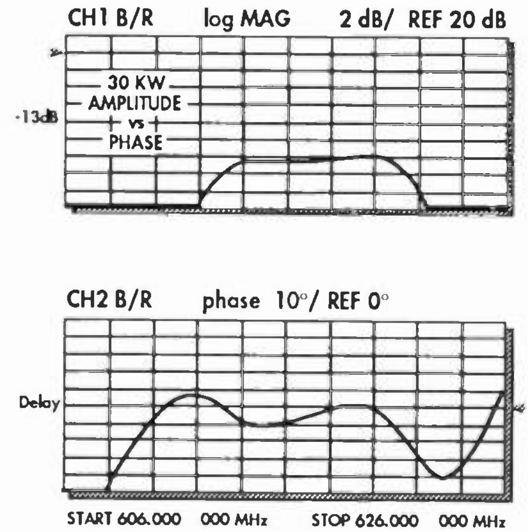

Figure 5

Figure 5 shows the results at 1.5 KW. The power gain variation across 6 MHz band was less than 0.2 dB and the group delay less than 12 degrees.

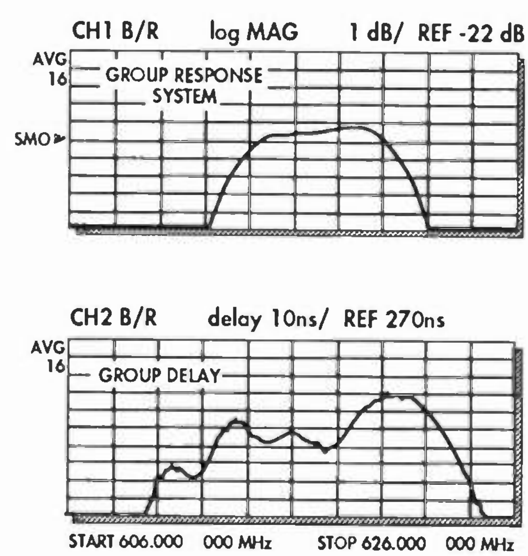

Figure 6

Figure 6 shows the results at 30 KW. The power gain variation across 6 MHz band was less than 1.0 dB and the group delay less than 35 nanoseconds. These measurements were all made without pre-correction. Readings were taken at the -1, -3, -6 and -10 dB down points from 30 KW peak power level. There were no significant changes.

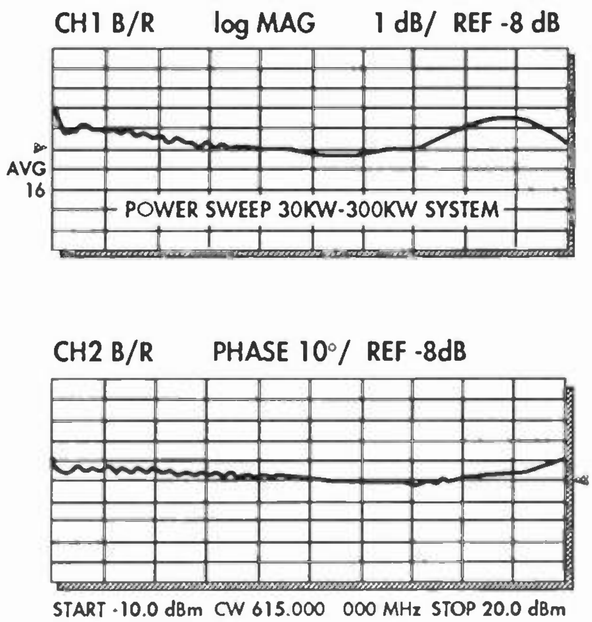

Figure 7

Figure 7 shows the amplitude and phase distortion from 30 Watts to 30 KW output. This power sweep from 30 Watts to 30 KW was taken in 1 MHz increments through the pass band. The results show a maximum of 2 dB linear gain variation and 18 degrees phase shift through this range of power. This figure shows the results at the center frequency. Again, there were insignificant changes at the other frequencies.

Figure 8 again shows the IOT compared to the Klystron transfer curve. The Klystron distortion shows up in the HDTV signal as out of band products. This means the tube must operate, with increased input DC power and the tube output power rating must be least 6 dB higher than the average power needed. This is needed to maintain linearity for the 8.5 to 10.1 dB over average power HDTV peak signals and to allow to use of a filter to remove out of band products.

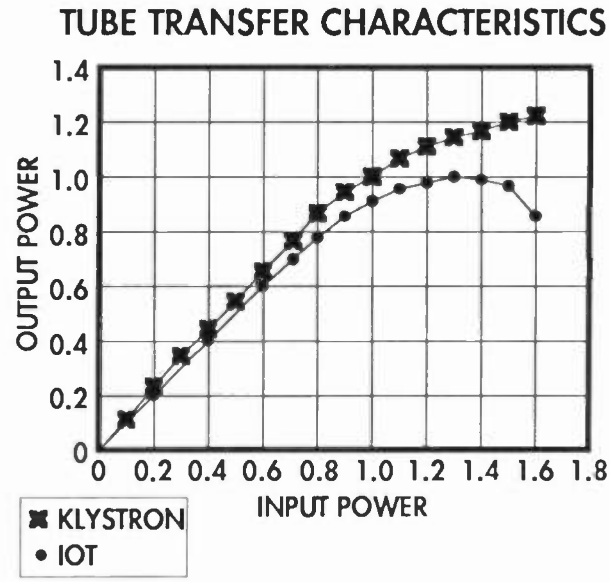

Figure 8

At TTC's plant, in cooperation with Zenith and Hewlett-Packard, we ran tests of the Zenith system on June 17 and 18, 1992, using our IOT transmitter. At our plant, we had the advantage of using the new HP 8990A peak power analyzer with a prototype HP 84815A sensor designed for analyzing signals like digital HDTV systems. The only difference between the transmitter we used and our standard IOT transmitter was the use of our prototype solid state driver. This new driver was used because it is wide-band, as it covers the entire UHF band, and because it is linear with intermod products, both in and out of band, performance of -62 dB without correction. Today there is no agreed specification for HDTV out of band products. Even without a pre-corrector we met all the known HDTV requirements except for the out-of-band NTSC spec of -60 dB. The out of band product results were - 42 dB below the pilot/ sync level and -33 dB below the average power output. However, MCI is supplying a filter for field tests for the HDTV system which should make the out-of-band products fall well within these parameters. The pre-corrector may also be adjusted to meet these specifications.

During the plant tests, the transmitter was made non- linear to distort the signal. Initially the input signal to the receiver was set to its noise threshold which is at the +15 dB Carrier/Noise level. At this level, any less signal will cause errors to be generated at the output of the receiver. When varying the distortion until the in and out of band products exceeded -22 dB, we still did not record errors. One method used to set pre-correction in the HDTV systems requires using a 16QAM analyzer and a 16QAM generator. Using HP's HDTV test signal, figure 9 shows a drawing of the 16QAM constellation on the HP 8981B Vector Modulation Analyzer.

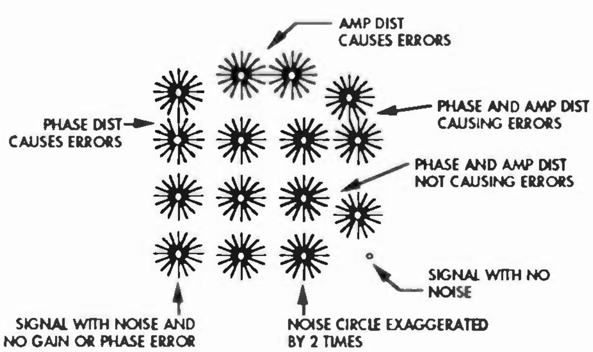

Figure 9

On this figure, the carrier to noise of +15 dB is shown at 2 times its normal deviation on the display for information purposes. The pre-corrector is adjusted for linearity and phase distortion which can be seen in the phase-plane or "Vector" display and can be quantified in terms of error rate compared to C/N degradation using software uncorrected. As you will note, there are distortion ranges which will not cause errors.

Figure 10 shows deviation from theoretical with various power levels. The lower curve shows the theoretical curve. The second shows the test equipment connected back to back. The next curve shows the transmitter at 6.1, 7.5 and 8 KW average power levels. The last curve shows a 2 dB increase in the deviation at the 9 KW power level. This still resulted in no bit errors in the system. On the Zenith signal, to evaluate compression of the peak signal, the HP peak power analyzer was used to measured peak power to average power ratios from 6 KW to 9 KW average power as measured on a Bird average power meter.

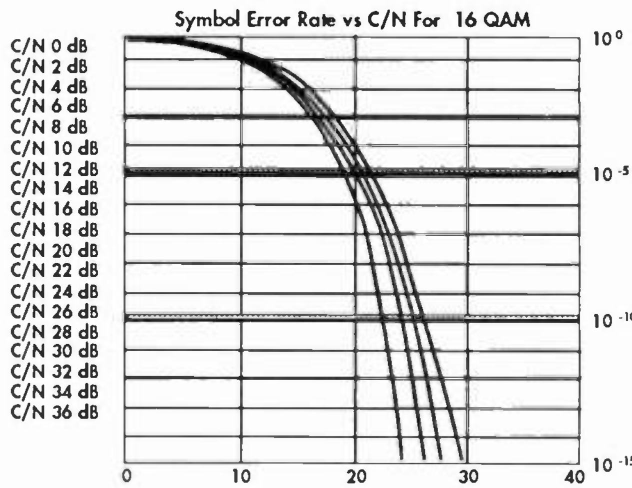

Figure 10

HP 8990A Peak Power Anal Data on the Zenith HDTV System June 18, 1992

Bird Aver Meter In kW	Peak In dB	Aver In dB	Peak/ Aver Diff	Note	Remarks
15.0	12.6	12.6	0	No Err	CW
6.1	14.4	7.1	7.3	No Err	Dig Sig (Xmitter)
7.5	15.0	8.2	6.8	No Err	Dig Sig (Xmitter)
8.0	15.62	8.69	6.93	No Err	Dig Sig (Xmitter)
9.0	15.78	9.00	6.78	No Err	Dig Sig (Xmitter)
9.0	12.65	5.87	6.78	No Err	Dig Sig (Driver)

Figure 11

Figure 11 shows the compression of the peak power compared to the average power, after 7.5 KW, was within the accuracy of the measuring equipment. The compression present was in the driver. The drivers requirement under this test was to provide enough power to allow the tube to reach 72 kw. During this test,the IOT tube's beam voltage was at 28 KV with a current drain of 1.5 amps or power input of 42 KVA for 9 KW average output or 43 KW peak power. This compares with an input power drain of 123 KVA for a standard Klystron or 72 KVA with an MSDC Klystron. With time and the final selection of the HDTV system, we believe correction techniques will allow further improvements in transmitter systems efficiency.

Here are our conclusions from these results and observations in the HDTV area. The HDTV systems we have examined are more robust than we initially expected. Our experience has shown that current transmitters can meet the requirements for HDTV but with poor efficiency compared to the IOT transmitter system. Standard or MSDC klystron transmitters will have to be run at a peak power level at least 6 dB higher than the average and pulsing will not be possible. With an efficient IOT transmitter, the equivalent beam power of a 60 KW peak transmitter will transmit at an average power of 15 KW. This will handle the peak signals without any errors being noted. We should point out that efficient, NTSC, HDTV-capable, IOT transmitters are available from TTC today. These transmitters will easily be able to provide complete digital HDTV coverage of major market areas with only the addition of a few minor components when HDTV transmission begins.

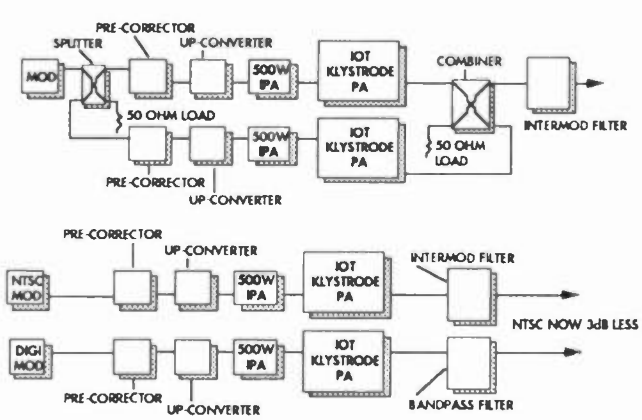

Figure 12

Figure 12 shows how an IOT transmitter system today, with the tubes power from 40 to 60 kw, can be modified to the required simulcast system tomorrow. The output filter would have to be purchased. All other components would be frequency modifiable.

Author's Note:

The author wishes to thank Carl Eilers of Zenith Electronics Corp. and Chris Pederson of Hewlett-Packard for their contribution on the tests made at our plant.

AN IOT TRANSMITTER FOR HDTV AND NTSC

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ABSTRACT:

This paper reviews the specification and requirements of a new Inductive Output Tube (IOT) powered transmitter to meet the needs of current NTSC and the transition to HDTV applications in UHF.

The characteristics of the transmitter are described in terms of its performance, power requirements, in band and out of band intermodulation products, linearity correctors and protection devices.

The use of broadband class AB amplifiers as drivers will also be covered with respect to linearity, correction, and efficiency.

INTRODUCTION

The future holds many exciting developments for HDTV terrestrial broadcast, and in turn, important decisions will have to be made by broadcasters on the choice of transmitter equipment to satisfy current and future needs.

Most of the UHF high power amplifiers available today and in use for NTSC will be usable for HDTV with varying degrees of performance and efficiency. Factors to be considered are peak power, average power, linearity, efficiency and conversion costs.

Sigma, a new transmitter from Harris Allied, has been designed to fulfil these requirements as well as providing low cost of ownership, high efficiency, ease of maintenance and a high level of manufacturing quality.

Figure 1 shows a 30kW Sigma IOT transmitter employing a single IOT power amplifier operating in combined amplification. This simple system forms the basic building block for transmitted powers up to 240kW employing up to six IOTs.

SYSTEM DESCRIPTION

The design of the new transmitter has focussed attention on system simplicity, ease of maintenance and use, and last but not least the ability to transition to HDTV when necessary.

The block diagram shows a main and standby exciter configuration followed by an RF corrector (one for each PA), a high gain broadband IPA, the Inductive Output Tube (IOT) and RF system.

EEV, the IOT manufacturer currently supplies two water cooled tubes; the IOT 7340 which can produce up to 40kW (visual only) and the IOT 7360 producing up to 60kW (visual only). Additionally, the smaller IOT 7340 tube is available in an air cooled version, and a lower power 20kW tube is planned. The PA cabinet described here will accommodate all tubes without modification, and all circuits and supplies are rated for the largest tube. The IPA, which is also rated for the worst case channel and maximum drive requirement, has an additional 2dB headroom. An optional switch can be supplied to redirect the IPA output to antenna as another level of redundancy.

EXCITER

This new exciter is a single pull-out drawer design with hinged IF modules, allowing maximum accessibility without the need for extender cards or leads.

Correction is provided for receiver group delay and group delay introduced by band limiting circuits, non linearity of the luminance and chrominance (differential gain) signals, differential phase, and ICPM.

As these circuits are wideband, cross-talk which occurs during combined visual and aural amplification is minimized, reducing the interference of the BTSC pilot signal by horizontal line syncs.

Video processing is applied to the incoming signal to improve the quality of transmitted video signals. This processing provides for the re-insertion and shaping of sync pulses and the limiting of video excursions beyond peak white.

Within the same drawer, a highly linear double balanced mixer is used for signal modulation followed by VSB shaping using a SAW filter. Up conversion to UHF uses double balanced mixers, followed by a channel filter and output amplifier. A standard exciter provides five UHF outputs capable of driving up to four IOT PAs plus a monitoring output.

When separate amplification of the visual and aural is required, then three additional modules are added to the exciter to provide an aural output for either an IOT or a solid state sound amplifier using the same advanced class AB technology used in the IPA.

The Exciter for HDTV

The pre-correction circuits currently being used for NTSC will be applicable when the changeover to HDTV is made. In making this conversion, a new digital modulator currently under development, will be substituted for the NTSC modulator.

The Power Amplifier Stage

The power amplifier cabinet is divided into 6 main sections, namely the RF corrector, IPA, IOT, control and monitoring, protection and cooling.

RF Corrector

Prior to the IPA and located within each PA cabinet is a patented wide band RF corrector. This corrector provides independent correction for each IOT and effectively makes the IPA transparent to the signal. At UHF, mid gray to white correction is generated in the RF corrector to compensate for the class AB non-linear amplifiers that follow. The circuit assists the IF carrier phase corrector at levels between mid gray and white, at the same time canceling third and fifth order intermodulation products.

Broadband Intermediate Power Amplifier (IPA)

A class AB IPA has been chosen for its efficiency and low component count using only 8 power transistors and producing enough power to drive the largest tube with a full 2dB of headroom. Our experience in UHF broadband correction of class AB amplifiers has resulted in an intermodulation product specification for the Sigma Transmitter of -60dB both in and out of band.

The IPA module is symmetrical, consisting of two parallel amplifier circuit boards each having two output ports driven from a common input.

The basic building block is a broadband bipolar transistor Class AB amplifier, which has a nominal black level gain of 10dB. These amplifiers are rated for television amplification at 150W peak sync.

Each half of the IPA has a class A driver with a gain of 34dB, feeding via splitters four class AB 150W transistors; these are paired to produce two outputs of about 280W peak sync. Each half also has a gain stabilizer to compensate for variations due to temperature, channel frequency and drive input level.

A patented dynamic linearity corrector is also used to compensate for device performance variances with average picture level. This is achieved by monitoring the average picture level and compensating the transistor bias.

The amplifiers are protected via a cut-back circuit within the gain stabilizer for high peak and average currents, reflected power, high heatsink temperatures and out of limit supply voltages.

For additional stability and protection the four outputs of the IPA are protected by circulators, the outputs of which feed a broadband star combiner, output coupler and low loss cable to the IOT.

The IPA is powered by two high efficiency three phase power supplies for added redundancy.

The IOT Power Amplifier

The PA cabinet shown will accommodate all the current and proposed IOT variants. As well as monitoring and control, the cabinet houses all the necessary magnet, filament, ion pump and bias supplies and protection circuits.

At the time of writing, the beam voltage recommended by EEV is 32kV for the IOT type 7360 to produce an output power of 60kW visual only, or 42kW visual (+4.2kW aural) in combined amplification. Higher beam voltages have been proposed in the order of 34-35kV to achieve a combined amplification power of 50kW visual (+5kW aural) but transmitter performance at these levels is yet to be proved.

Dual tube (Figure 2) with combined amplification of visual and aural provides very good system redundancy as well as a solution to HDTV re-configuration at a future date.

The redundancy and transitional HDTV system re-configuration features are likely to be a popular choice by broadcasters. At 32kV, tube types IOT 7360 and IOT 7340 provide an absolute peak power of 73kW and 56kW respectively for combined amplification. Using these two absolute peak power ratings, the SIGMA Transmitter will produce operational powers of:

	Type IOT 7360	Type IOT 7340
Visual Only	60kW	40kW
Combined Amp		
Visual	42kW	32kW
Aural	4.2kW	3.2kW

For example a two tube transmitter working in combined amplification would provide total output powers, after system losses of:

	Visual Power	Aural Power
2 x IOT 7360	80kW	8kW
2 x IOT 7340	60kW	6kW

The following calculation, based on data released by the tube manufacturer, confirms the maximum power ratings. The calculation is for the IOT 7360 using a beam voltage of 32kV and at mid channel UHF. In common amplification, the RF peak envelope power is determined by the peak RF voltage, which is actually the sum of the visual and aural voltages and calculated as follows:

$\text{Power} = \frac{E^2}{R}$ Therefore $E^2 = \text{Power} \times R$

Where R = 50 Ohms

Using the maximum power rating for IOT 7360 in combined amplification.

Visual	Aural
$E^2 = 42{,}000 \times 50$	$E^2 = 4{,}200 \times 50$
$E^2 = 2{,}100{,}000$	$E^2 = 2{,}100{,}00$
E = 1449.14 volts	E = 458.26 volts

We now add the voltages together to get the combined peak voltage

Esum = 1449.14 + 458.26

Esum = 1907.4

$\text{Peak envelope power} = \frac{E^2}{R}$

$$\text{PEP} = \frac{1907.4^2}{50}$$

Absolute max PEP = 72.76kW

Crowbar Protection

All IOTs have to be protected from damage caused by internal arcs, and this is provided by the beam voltage power supply crowbar circuit. A device must sense a rapid increase in current and interrupt the beam current before damage is caused. This action is accomplished with a beam current detector and a crowbar in parallel with the IOT. Once the crowbar has fired, high speed vacuum switches (10mS) remove the three phase mains supply.

Figure 3 shows a typical beam supply and crowbar circuit.

High speed reliable operation is essential for this circuit and several different devices could be used. Most can be discounted because of the HV requirement, cost or complexity, leaving a choice between the thyratron and the triggered spark gap.

Triggered Spark Gap Crowbars

These devices are of relative low cost but their reliability in high voltage applications is not ideal for the following reasons:

Beam voltages of 32kV determine the spark gap rating and maximum hold-off voltage. The spark gap has a dynamic range of 2:1 and therefore the spark gap requires a minimum breakdown voltage of 16kV to operate.

Under an IOT arc condition, the voltage across the crowbar, V2, drops to about half of V1 (16kV, ignoring any stray inductance), the over current is detected and the spark gap fires.

The spark gap is an erosive device, and with age, the breakdown voltage of the spark gap increases, increasing the risk of failure to fire.

This can be overcome by reducing the value of R1 to increase the breakdown voltage V2 in the arc condition. This has the effect of increasing the current through the spark gap when fired, further reducing its life and efficiency.

Using a spark gap with a hold-off voltage of 35kV will result in the lack of performance as mentioned above and a spark gap of less than 32kV cannot be used, therefore different spark gaps have to be used for different operating beam voltages.

A further limitation on the device is the requirement of a high voltage trigger circuit and a step-up pulse transformer. This introduces a small delay into the crowbar protection circuit allowing more energy to be dissipated in the IOT with a greater possibility of damage.

Thyratron Crowbars

The thyratron provides greater reliability in crowbar protection. It has dynamic range of 1kV to 35kV and only requires a few hundred volts to trigger the device. This being provided by the current sensing transformer rather than a separate trigger circuit.

The grid of the thyratron can be monitored to check its serviceability and interlocked into the transmitter control circuits.

The thyratron has a switching delay of approximately one microsecond providing the fastest possible removal of the beam supply current.

Experience has shown that some spark gaps are prone to random triggering, often at transmitter turn on. Testing at Harris Allied has proved that the thyratron protection circuit fires only as the result of an actual IOT arc.

Therefore, despite the small cost advantage of the triggered spark gap thyratron protection is preferred because of its speed and reliability.

Control and Monitoring

The control and monitoring units have been ergonomically designed for ease of use. They are fully solid state, containing no electro-mechanical relays. Use of embedded microprocessors has been avoided but external interfaces have been provided to meet the requirements of many commercially available remote control systems.

Each PA cabinet has its own independent control, monitoring and remote interface allowing it to be maintained as a separate entity. The exciter rack, which can accommodate dual exciters, is also equipped with a system control panel, system monitoring, power metering and full remote control interfaces. This rack can control up to six PA cubicles.

Cooling

The IOT collector can be cooled using either water or air. The cavities and IPA are air cooled with slow speed fans to ensure an ambient noise specification of 65dBA or less.

External to the cabinets various water cooling systems are available using redundant pumps and heat exchangers with either distilled water or glycol mixture. These can be individually designed to meet site requirements.

The RF System

A combined amplification visual and aural transmitter will require a RF output filter to reduce intermodulation products but does not need any visual/aural diplexer.

Traditionally out-of-band intermodulation product filters have consisted of a series loosely coupled short circuited stubs on the output transmission line which reflect energy at the main product frequencies. This had the disadvantage that the energy was reflected back to the power amplifier where it contributed to the generation of further products.

Sigma transmitter systems employ balanced absorptive filters. These filtering elements are placed in a hybrid ring between two couplers. The transmitter output is connected to one coupler and the antenna to the other. The remaining two ports are terminated with loads. With this arrangement intermodulation product energy is diverted into a load and does not cause further intermodulation in the power amplifier.

Transmitter Systems

IOTs offer a level of efficiency only marginally exceeded by MSDC type klystrons in separate amplification, but in combined amplification the IOT becomes the more effective device. This combined with simpler cooling and single voltage beam supplies, no pulser and ease of maintenance, IOTs are a good choice for current NTSC and future HDTV applications. The operation of high power amplifiers in combined mode has only been generally used for reserve operation in the USA. Harris Allied, however, has supplied many common amplification systems in Europe for over 25 years, and have gained the necessary experience to design the stable pre-correction systems needed for this mode of operation. Figure 2 show a typical 60kW, combined amplification, transmitter block diagram.

Combined amplification has its main advantage in system redundancy. In a 60kw system, two identical amplifiers each of 30kw peak sync are employed. The failure of an IOT amplifier results in only a 3dB drop in output power after combiner adjustment. Duplication of the beam supplies, cooling systems and exciters operating in a main and standby configuration further enhance the system redundancy.

Transition to HDTV

The flexibility of the Harris Allied exciter and a dual tube combined amplification transmitter, provides an excellent system for re-configuration and upgrade to HDTV.

By dedicating one PA to HDTV, the remaining PA can be fitted with a larger IOT to provide increased NTSC output power. HDTV transition can be achieved with minimum cost.

Performance

HDTV systems are described as digital systems. This terminology applies to the digital signal processing used on the base band audio and video signals. Once modulated on to an RF carrier it becomes analog and will suffer the same distortions as a NTSC signal as it is frequency converted, band limited, amplified and transmitted.

This implies that the NTSC transmitter, when properly set up can successfully transmit an HDTV signal. What needs to be examined are the key NTSC parameters and their influence upon the HDTV signals.

NTSC Power Consumption and Efficiency

A combined, single tube, 30kW, Sigma transmitter (delivering 32+3.2 kW at the tube flange) has a total transmission plant power consumption of 43.65 kW when transmitting an average program. This figure, which includes aural, exciter, drive amplifier etc., gives a plant FOM of 68.7. This is made up from:

	Power
Beam (mid gray)	36.9kW
Drive amplifier	1.5kW
Exciter, logic, heater and focus.	0.63kW

	39.03kW
Plus power supply conversion efficiency at 95%	40.98kW
Cooling plant	2.67kW

	43.65kW

For this measurement, average program has been taken to be a true mid gray of 43.75 %, i.e., half way between peak white at 12.5% and blanking at 75%.

Care must be exercised when comparing transmitter efficiencies. When the method set out in EIA-508 Sect 1-12 is used, a 50% APL or a gray of 37.5% is specified. As this is nearer white, efficiency will appear to be better.

If the Sigma transmitter is measured under EIA-508 conditions, the overall plant consumption decreases to 39.03kW returning a plant FOM of 76.8.

The power consumption for a combined 60kW transmitter would depend on the cooling system employed and roughly be a little less than double the above power calculation.

HDTV POWER LEVELS

In examining the four proposed digital HDTV systems, the main problem is in determining Advanced Television (ATV) transmitter power levels. At this time, there is a high level of uncertainty that exists due to the lack of accurate definition of the various parameters. For example, the ATV transmitter power level referred to in the various ATV proponent system disclosure statements, is average power, although not explicitly stated as such, while the customary NTSC power level is defined as peak of sync power. It is the peak power that establishes the transmitter power rating.

This has lead to a considerable amount of confusion in determining the actual transmitter power for ATV. The average power for ATV is small and very favorable while the peak values were quite high and nearly equal to NTSC peak values. The average power is one way to measure ATV power, but when trying to determine transmitter power ratings, the peaks must be accounted for and the analysis begins to get complex. Table 1 shows an estimate of the proponent peak to average ratios and referenced to NTSC.

Each proponent has a different peak to average power ratio which leads to significant differences in the required transmitter power. All of this is then superimposed on the various TV station system variables, i.e., antenna gain, antenna height, transmission line/diplexer losses, and various signal to noise ratios to establish the required service area contours. Also, as of this writing, the final ATV proponent system has not been selected yet. This can lead to numerous planning scenarios where all of the system variables are required to arrive at a reasonably accurate ATV transmitter power calculation.

Therefore, there is no firm and fast way to accurately calculate ATV transmitter power levels until the areas of uncertainty are clearly defined.

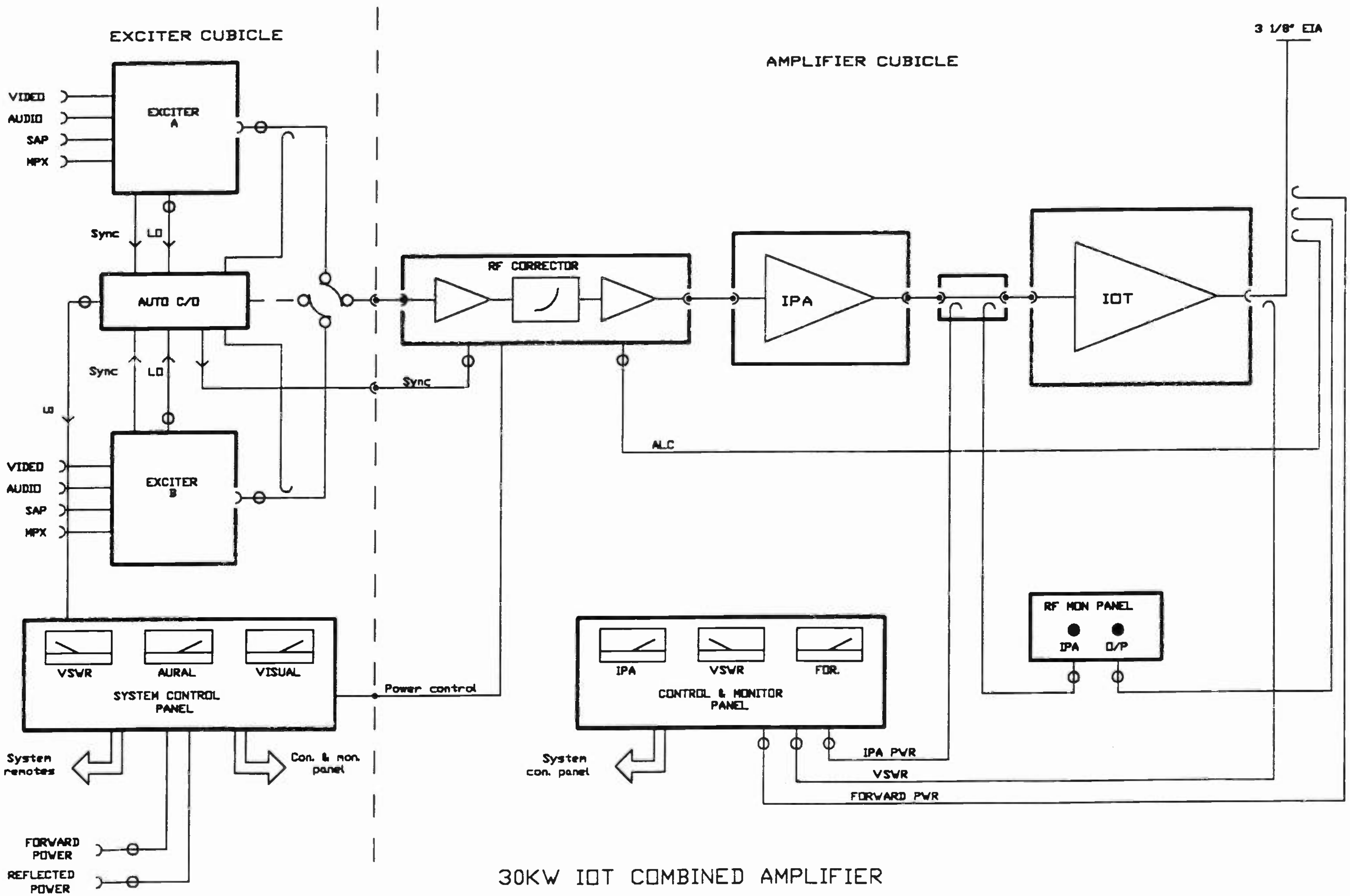
CONCLUSION

The paper has outlined the features and performance required of an IOT transmitter and transmitter system designs for NTSC and HDTV. The combined amplification of visual and aural signals and the use of new correctors and class AB drivers to provide new improved levels of linearity and efficiency.

ACKNOWLEDGEMENTS

The author would like to thank the management of Harris Allied for permission to publish this paper and his colleagues at Harris Allied, Quincy, USA and Harris Allied, Cambridge, UK. for their assistance in its preparation.

The author would also like to thank English Electric Valve Ltd (EEV) for their tube and protection data.



30KW IOT COMBINED AMPLIFIER

FIGURE 1

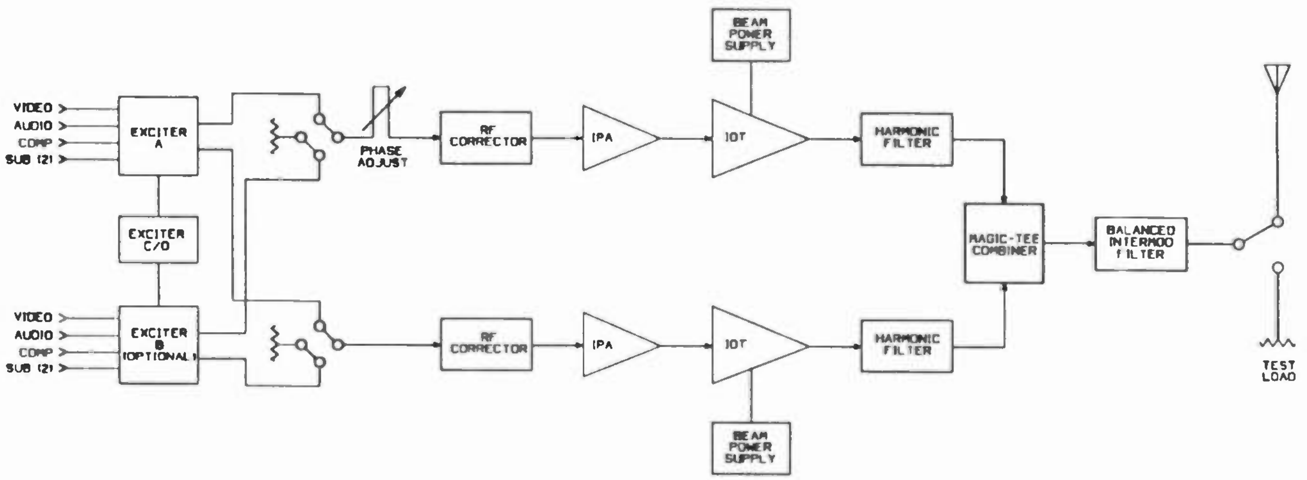


Figure 2 - 60kW IOT TRANSMITTER RF BLOCK DIAGRAM

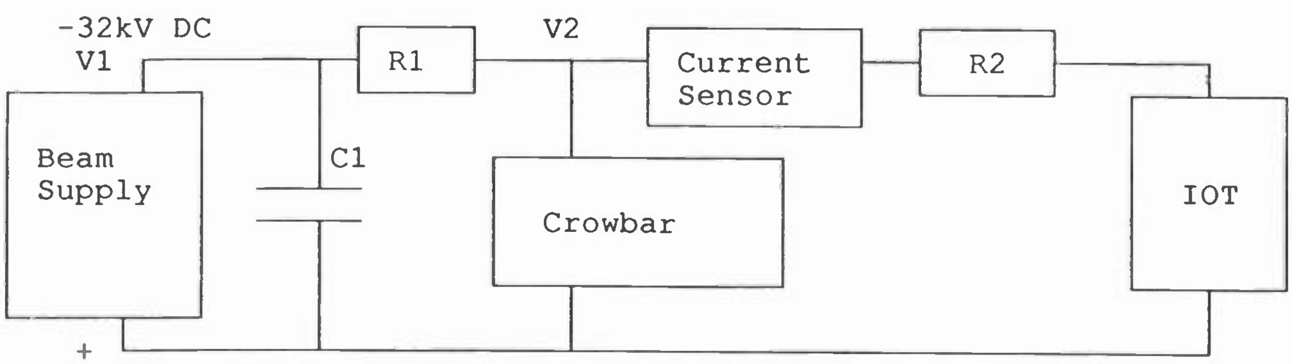


Figure 3

	Transmitter Reference Power NTSC	Digital ATV Power for Equivalent Coverage			
		Digicipher	MIT	Zenith	Sarnoff
Peak	120kW	27kW	34kW	43kW	67kW
ERP	2,500kW				
Average	38kW	7kW	7kW	7kW	7kW

Table 1

IEEE BROADCAST TECHNOLOGY SOCIETY
BROADCAST STANDARDS: WHY, HOW AND WHEN?*

Monday, April 19, 1993

Moderator:
Thomas E. Hankinson, Capital Cities/ABC Inc.
Philadelphia, Pennsylvania

This session addresses the subject of broadcast standards in the USA and worldwide. It provides a glimpse into who creates these standards, how they go about it, when standards are required and how they are used.

Presentations cover an overview of standards development, the interrelation of national and international organizations, and the allocation of standards responsibility among the various interested parties.

Three examples of standards are explored in detail. One is Ghost Canceling in Television, another is the IEEE-488 Bus standard, and a third is a standard developed for radio.

AN OVERVIEW OF STANDARDS DEVELOPMENT
George Hanover
Electronic Industries Association
Washington, District of Columbia

NATIONAL AND INTERNATIONAL APPROACHES TO BROADCAST STANDARDS
Stan Baron
NBC Television Network
New York, New York

ALLOCATION OF STANDARDS RESPONSIBILITY
James C. McKinney
Advanced Television Systems Committee
Washington, District of Columbia

EXAMPLES OF EXISTING STANDARDS:
NRSC RBDS RADIO STANDARDS
Almon Clegg
Consultant
Rockaway, New Jersey

GHOST CANCELING - ATSC
Tony Uyttendaele
Capital Cities/ABC
New York, New York

IEEE-488 BUS
Joseph E. Mueller
Hewlett Packard
Loveland, Colorado

*Papers for this session were not available at the time of publication.

BROADCAST AND RELATED TECHNOLOGY STANDARDS ORGANIZATIONS ADDRESS, TELEPHONE AND TELEFAX LISTING

January, 1993

American National Standards Institute (ANSI)
655 15th Street NW
Washington, DC 20005
Phone 202-639-4090
FAX 202-628-1886

Advanced Television Systems Committee (ATSC)
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Washington, DC 20006
Phone 202-828-3130
FAX 202-828-3131

Audio Engineering Society (AES)
60 East 42nd Street
New York, NY 10165
Phone 212-661-8528
FAX 212-682-0477

Cable Television Laboratories, Inc. (CableLabs)
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FAX 303-939-9189

Electronic Industries Association (EIA)
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European Broadcasting Union (EBU)
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Federal Communications Commission (FCC)
Mass Media Bureau, Engineering Policy Branch
1919 M Street NW, MS 1800D3
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FAX 202-653-9659

Institute of Electrical & Electronics Engineers, Inc. (IEEE)
Broadcast Technology Society (BTS)
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International Radio Consultative Committee (CCIR)
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International Standards Organization (ISO)
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National Cable Television Association (NCTA)
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U.S. Department of Commerce
Gaithersburg, MD 20899
Phone 301-975-2000
FAX 301-975-2128

Society of Cable Television Engineers, Inc. (SCTE)
669 Exton Commons
Exton, PA 19341
Phone 215-363-6888
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Society of Motion Picture & Television Engineers (SMPTE)
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White Plains, NY 10607
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FAX 914-761-3116

IEEE BROADCAST TECHNOLOGY SOCIETY
BROADCAST STANDARDS: ROADBLOCKS OR GUIDEPOSTS?

Monday, April 19, 1993

Moderator:
Alan Godber, Independent Consultant, Milltown, New Jersey

This session addresses the subject of broadcast standards and when they are a help or when they may be a hindrance to the industry. It explores how these standards have been created, it explores the successes and the failures, and it looks at this from the point of view of the USA and from a worldwide point of view. Radio and television broadcasters, professional and consumer equipment manufacturers points of view are heard. The future of broadcast standards is also explored and predicted.

A panel discussion also drawing on other viewpoints concludes the session.

APPLICATIONS OF STANOAROS INTERNATIONALLY
Ken Davies
Canadian Broadcasting Company
Montreal, Canada

RADIO BROAOCASTERS' PERSPECTIVES
Al Resnick
Capital Cities/ABC
New York, New York

TV BROADCASTERS' PERSPECTIVES
Carl Girod
Public Broadcasting Service
Alexandria, Virginia

BROAOCAST EQUIPMENT MANUFACTURERS' VIEWPOINT
Larry Thorpe
Sony Advanced Systems
Montvale, New Jersey

CONSUMER ELECTRONICS EQUIPMENT MANUFACTURERS' VIEWPOINT
Joseph Donahue
Thomson Consumer Electronics, Inc
Washington, District of Columbia

Panelists:
All Participants and Gerry Berman, Voice of America, Washington, District of Columbia, Bill Hassinger, Federal Communications Commission, Washington, District of Columbia, Ike Blonder, Blonder Tongue, U.S. IEC representative, Morganville, New Jersey

*Papers for this session were not available at the time of publication.

SBE DAY
DEALING WITH RF INTERFERENCE

Tuesday, April 20, 1993

Moderator:
John Battison, P.E., Consulting Engineer, Canton, Ohio

STL SYSTEMS: HORROR STORIES AND FIXES
George Whitaker
Practical Radio Communications
Arlington, Texas

DESIGNING A BULLET-PROOF RPU SYSTEM
Paul Montoya
Broadcast Services of Colorado, Inc.
Denver, Colorado

RESOLVING BROADCAST-RELATED INTERFERENCE TO CONSUMER ELECTRONICS EQUIPMENT
Nathan Hamilton
Hammett & Edison, Inc.
Consulting Engineers
San Francisco, California

***FACILITY GROUNDING: PRINCIPLES AND PRACTICES FOR REDUCING RFI**
Birken Olson
Current Technology
Richardson, Texas

FREQUENCY COORDINATION: THE BEST SOLUTION TO INTERFERENCE
Richard A. Rudman
Chairman, SBE National Frequency Task Force
Indianapolis, Indiana

***GETTING HELP WITH INTERFERENCE: HELLO FCC**
Richard Smith
Federal Communications Commission
Washington, District of Columbia

***MW RERADIATION FROM SIMPLE VERTICAL STRUCTURES**
Karl D. Lahm, P.E.
Consulting Engineer
Fairfax, Virginia

*Paper not available at the time of publication.

STL SYSTEMS: HORROR STORIES AND FIXES

George Whitaker
Practical Radio Communications
Arlington, Texas

Interference between STL's in metropolitan areas is, unfortunately, a relatively common occurrence. Such interference usually comes about when one station constructs, or alters, a link. Through frequency coordination these incidents have been kept to a minimum and usually it is possible to identify the interfering station quickly and corrective action is usually pretty standard. However, in this paper we will take a look at some cases where the causes, and culprits, were not so readily apparent.

In our first case we have a long-existing link operated by KSSA in Dallas that suddenly was picking up what sounded like a video carrier.

The KSSA link has the transmit end in downtown Dallas utilizing an eight-foot dish pointing NNE. The receive end is an eight-foot dish on a 400-foot tower in the suburb of McKinney. It is a 34 mile hop with a Scala preamp giving additional help to the receiver.

In early summer of '92 KSSA began getting interference that sounded like a video carrier. No real attempt was made to find the source. A cavity was purchased and placed in front of the preamp and it cleared the problem.

However, in September the problem reoccurred and the interfering signal was about equal strength with the STL carrier. KSSA rents tower space to several paging companies and the first thought was that one of them had a problem that was manifesting itself in the STL. However, when a spectrum analyzer was brought out to the site, it became obvious that the problem signal was originating elsewhere.

By this time several days had elapsed and it was also obvious that powerful help was needed. The FCC field office in Dallas was called and given what information was available.

Within an hour, Ray Turner of the local office had produced the following information and gotten the offending transmitter turned off.

It seems that Northern Telecom of Richardson, a town on a direct path between KSSA's downtown location and their McKinney location, had been given an experimental license by the FCC that allowed them to bring up a transmitter anywhere between 946 and 949 mHz.

Northern Telecom had not contacted the local frequency coordination committee but had used a "frequency search" company which had produced that range of frequencies as being "available". The FCC had granted them the experimental license with the proviso that "no interference to existing facilities be caused".

Unfortunately, no one at Northern Telecom listened to, or looked at with an analyzer, the frequency chosen from the band allotted. They brought up a 16 watt transmitter on 946.6 mHz utilizing an omnidirectional antenna. This was the signal that originally interfered with the KSSA STL. KSSA was blowing right across their site on 946.5 mHz.

Northern Telecom was sending data with their installation and apparently did not have an audible signal to hear KSSA, which should have shown up in their receiver also. The data rate was such that KSSA thought it was a video carrier.

The Northern Telecom transmit site is approximately halfway between the KSSA transmit and receive points. This meant that, by the time it got to McKinney, the bandpass filter installed by KSSA was able to reduce the interfering signal to a negligible value.

Then, in September, Northern Telecom came on with their second transmitter on 946.6. This one was a 32 watt unit, again utilizing an omnidirectional antenna. It completely wiped KSSA out. The bandpass filter could not handle this kind of signal strength only .1 mHz removed.

The outcome was that Northern Telecom became aware that broadcasters all over the Metroplex were operating in that band and they began working with the local frequency coordinator. It is my understanding that they moved to frequencies that were being used to go SSW of Dallas, installed directional antennas, and reduced power to 1 watt at the antenna. So far it appears to be working.

The lesson to be learned here is that you can not depend on the regulatory agencies to coordinate frequencies. A blanket license was granted for a block of spectrum that was in heavy use in the Dallas area. Likewise the "frequency search" company looked at non-broadcast use only and decided that, if the telephone companies weren't using it, and the government wasn't using it, it must be vacant. Be aware that you can get "licensed" interference.

Our next case history takes us to Indianapolis and WFMS. Here the studios were downtown and the transmitter site was on the southeast side of the city. As STL's go, this link had never been the best although noise spec could be met. However, there was not a lot of room for fade, so a Microwave Associates active splitter was used to divide the signal between the active and stand-by receiver. This unit had unity gain and a bandpass of 850 to 960 mHz. The link had operated well for several years.

Then, about 45 db down, came audio and noise. The chief engineer at WFMS called the local coordinator with the obvious question, "Is anybody doing anything?" The answer was "No, nothing new; no changes."

Although basically unintelligible, the fact that the interference sounded like program audio indicated that it almost had to be a station somewhere.

Then someone recalled that channel 59 had just begun program tests. Could a UHF TV station on the other side of town be causing the interference? Yes.

Channel 59's transmitter was on the northeast side of Indianapolis. WFMS had their receive site on the southeast side of town with the studios in the northeast quadrant. This meant that channel 59 was several degrees off the path. However, 5 megawatts is a lot of power and the WFMS dish was getting enough of it to drive the active splitter into distortion and it began to allow the demodulation of the TV audio along with the station's signal. This interference wasn't an STL of any kind. It was a main carrier. Therefore, frequency coordination had no place in the equation.

It is possible that a band-pass filter would have corrected the problem and still allowed a "no loss" splitter. Then, of course, you would have had the loss in the cavity. So, the active splitter was replaced with a passive device and the receiver would still make spec on noise with a signal of about 600 microvolts.

Again, it's not always another STL that's causing the interference.

Next, we find that the end result of interference was actually a gain. For this case history we go to Kalispell, Montana, where KALS has a receive site on Blacktail Mountain. This link had been in operation for about 6 years on 950.0 with excellent reliability, using a Moseley PCL 505 set.

During the fall of 1988, another lessee on the mountain began testing a new point-to-point system using frequencies in the 952-953 mHz range, causing interference to KALS. The digital data on the point-to-point system raised the KALS noise floor to 42db below 100%. And worse, the interference was occasionally strong enough on the STL receiver subcarrier that it interfered with transmitter control.

The company causing the interference was very cooperative, bringing in spectrum analyzers and a communications monitor to characterize the problem. Sure enough, there was enough signal on 950.000 mHz getting into the STL receive antenna to affect the audio noise floor of the receiver.

Their first attempt at solving the problem was to upgrade the STL antennas on both ends of the system. They went from six-foot truncated "short haul" type antennas to eight-foot full circular grid dishes. The thinking was that they would add a few db system gain which would help override the interference; and, would improve the side-lobe rejection in the direction of the offending transmit dishes. The other company's personnel agreed.

Alas, good ideas don't always yield the desired results. Wrestling the new dishes into place at 7 below zero in 2 feet of snow improved the received signal, lowered the noise floor, took care of the transmitter control problem, and gave everybody frozen fingertips. But it was not the total solution.

The KALS chief engineer thought that cavities might be of benefit and he proposed this to the interfering company. They weren't real enthused about trying something else, since they had just paid for the new dishes. Instead, they engaged a communications system engineer to analyze the problem and recommend a solution. After several weeks of study, he called with his proposed solution--try cavities.

So, armed with a new triple cavity assembly, the KALS crew headed back up the mountain. The cavities had 1/2 db loops in each port. Therefore, they expected to see a 1 1/2 to 2 db drop in received signal strength. However, when they brought the system back on line with the cavities in place, they were surprised to see a noticeable INCREASE in signal strength. The interference was completely eliminated and everyone was happy again. The conclusion was that the receiver had been desensitized by off frequency signals. The cavity filtered out the garbage allowing only "on frequency" signals into the receiver front end.

This next case history is being included because of the unique "fix" that was applied. It gives rise to thinking of solutions in a different perspective.

WAJL in Orlando, Florida, had just installed their new transmit dish on a tower that allowed the signal to just clear the top of a movie theatre across the street and give them line-of-sight. It came up just fine and it appeared that all was settled.

Then came a call from WJYO. The WYJO receive point was approximately 165 degrees off the WAJL path. So the interference was a case of a northbound signal getting into a southbound signal's receiver. There didn't appear to be any reason for that kind of radiation off the back-side of the WAJL dish.

Then, someone suggested that maybe the

signal was hitting the metal air conditioner on the top of the movie theatre and bouncing back to WJYO. The angle appeared to be about right. The WAJL engineering crew got permission from the theatre owner and constructed a sheetmetal lean-to on the side of the air conditioner. This served to deflect the bounced signal straight up into the air instead of back at WJYO.

It would have been almost impossible to stop the signal from bouncing off the air conditioner, so, they just bounced it off where it wouldn't hurt anything.

Our next case involves an example of what can happen when the "should be obvious" answer isn't.

KLIF, licensed to Dallas, had their studios in the suburb of Arlington, right on the side of I-30. To the southeast was Cedar Hill where the transmitter was located. The STL dish was on a 40 foot tower beside the studio building looking across I-30 toward Cedar Hill.

One day it became obvious that the STL signal had dropped to a fraction of its original level. The signal didn't seem to have an interfering signal mixed with it, it just wasn't there.

The usual things were checked. Towers were climbed and lines inspected, connectors and feedhorns checked for water, and transmitters and receivers were checked over from stem to stern. Then the dishes were realigned both horizontally and vertically. Nothing made any improvement.

About this time a new chief engineer arrived on the scene and began to learn about the problem. He talked with the engineering staff about all that had been done. Then he climbed the tower to the transmit dish to have a look for himself.

Looking across I-30 through the dish he found that he was looking directly into a billboard on the other side of the highway.

That billboard had been over there on that side of the highway when the link was built, but it wasn't in front of the dish then, the engineering staff assured him. And, they were right. The city of Arlington had passed a new ordinance allowing larger billboards on the interstates and the billboard company had enlarged the billboard to the new size limit. The billboard that had always been there just got bigger without anyone at the station noticing. It also just happened to now be in front of their STL dish.

It took over six months of dealing with the bureaucracy but they finally got a zoning variance to take their 40 foot tower on up to 100 feet and solve the problem.

I'll let you draw your own conclusions about what you can learn from a situation like this, where it took several actions to create the problem, it took a long time to find because it was too simple, and then the bureaucracy made a simple fix difficult.

In conclusion I would like to review just a few of the basic methods available for dealing with interference and preventing it in the first place.

When there is more than one transmitter on a tower, all of the transmitters should have isolators (circulators) on them to prevent intermodulation. An isolator is like an RF diode that keeps stray RF from reradiating back into the transmitter and causing the intermod. It has been found that, when a station is having trouble with a receiver on an antenna farm, it is usually the result of bad installation practices at one or more of the transmitters.

One of the biggest helps to cleaning up receive interference is the use of tuned cavities. In a majority of interference cases a simple bandpass cavity in front of the receiver will eliminate the problem. In extreme cases you may have to use band pass/band reject cavities to further isolate the receiver. Tuned cavities will filter out any other frequencies that may cause the receiver sensitivity to suffer. Even though a cavity, either band pass or band reject, will have some inherent loss, if there is interference causing de-sensing of the receiver, the cavity will actually increase the signal level providing better audio.

And finally, make sure that your audio wiring is properly installed. Many RF installations actually suffer from interference that is coming in through the audio lines.

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DESIGNING A BULLET-PROOF RPU SYSTEM

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Designing a bullet proof RPU system is virtually impossible... Designing a bullet resistant system however just takes a little careful planning and attention to details. Creativity is our best key to unlocking the secrets of "Remote" engineering. Sharing tips, hints and just common sense ideas in this art can be our encyclopedia to new innovations in RPU system design. We should always explore alternatives means to get a signal from "round the corner" or "across the globe" locations back to the studio.

I have never tried to present myself as an expert on anything when it comes to broadcast engineering because I am always proven wrong by someone brighter or with more experience. What I want to share with you now are observations and information picked up over the last few years from many situations in my day-to-day work and from other engineers in the trenches.

Normally when we think of a Remote Pickup (RPU) system we think of the "Marti" type broadcasts from the shopping mall. I would like to explore any and all of the means possible to get the best audio signal we can back to the broadcast facility. Today, if we relied only on the traditional means of RPU we would be selling ourselves short given the many new innovations and technologies on the market today.

Let's first look at the different avenues available in getting a signal around including:

- 161 Mhz RPU
- 450 Mhz. RPU
- Equalized Broadcast Lines
- Dial-up (Tip and Ring) Lines
- "Switch 56" digital line technology
- VSAT satellite transmission

161 and 450 Mhz. RPU

RPU frequencies in the 161 Mhz. and 450 Mhz. band have been the traditional way of getting signals back to the studio... except for maybe a dial-up phone and coupler, but let's stick to "grownup radio". Frequency congestion in many urban and sometimes medium markets cities has made it difficult to use this method. It is still the preferred way to broadcast remote in most parts of the country.

These frequencies are becoming more of a possibility again as stations are being combined through LMA's and duopoly purchases. A few years ago it was virtually impossible to use a 450 or 161 frequency (coordination or not). It seemed that every station in the market wanted to broadcast on the same Saturday morning you promised your sales manager you'd get the morning show on live from Slick Willies Used Car Emporium. Don't get me wrong, I know it's still congested in many markets, but overall the crunch is not bad. As things settle into place in the overall radio marketplace it will be likely that these frequencies will be used quite a bit again.

Thanks to new technologies though it will be possible to get quality audio down more narrow "pipes". TFT, Inc. in Santa Clara, CA is testing units that push digital audio down a typical 50 Khz. "R" Channel. Between Digital Signal Processing (DSP) chips and new compression schemes incredible spectrum utilization may soon be a reality.

Equalized Broadcast Lines

In the 60's and 70's equalized broadcast loops were a popular means of getting programs back to the studio. This was however before Judge Green; when we had only one phone company. Installation and reoccurring costs for our 5, 8 and 12 Khz. audio bandwidth circuits went through the roof and forced us to look at alternative transmission systems.

Some prices have come down but typical installation charges are still between $400 and $1200. Local operating companies have found that broadcast service is not as lucrative as many other services they provide so they make sure their costs are covered and then some.

Dialup Lines

Dialup lines are still very cost effective but who wants an audio bandwidth of 300 - 3200 hz.? Well, a few years ago a few companies like Comrex and Gentner started messing around with the audio sent down these "tip and ring" lines. Through frequency shifting, equalization, compression and expansion they have arranged to make dial up remotes with quality a desirable alternative to traditional systems. They found that by shifting audio frequencies up when sent out onto the phone line and shifting them back on the receive end they could "warm" up the bass response enough to make it attractive.

Then someone had the great idea of dividing up the audio spectrum and sliding different slivers down different phone lines to the studio and recombining them on the other end. They did it first with two lines and now even three.

I most recently had the opportunity to work with the Comrex 3XP/3XR frequency extender that can tie in three lines for audio response from 50 Hz. to 8 Khz. The 3XP encoder unit splits the audio into three sections; low, medium and high. The audio is then sent to a noise reduction system that further splits the bands, compresses them and then recombined them within each of the three bands. They are then frequency shifted and fed down a standard 1FB phone line.

The 3XR decoder reverses the process by downshifting the audio, expanding the bands and then recombining them to provide incredible audio.

Of course most of the time this means ordering in additional lines, as most fast food restaurants are lucky to provide you with one line, much less three.

"Switch 56" Lines

Another way of shoving audio down copper telephone wires is digitally. Through digital encoding and compression, your local operating company and cooperating long lines companies can provide your with high quality service pretty much coast to coast. You pay for the service on a per minute basis but the typical three hour remote would cost you a little over $45. Installation can vary quite a bit, but once you get a line installed to the main office it's just a matter of connecting up to your source destination.

Encoding and decoding equipment you purchase for each end is not cheap either, but sometimes it's easier to swallow in that one time capital purchase. It will be interesting to watch this transmission system in the future as it settles in and hopefully encoding/decoding equipment prices come down.

VSAT Satellite Transmission

Satellite was once a transmission means only for the people with unlimited budgets and large trucks. VSAT has brought satellite for remote broadcast and it doesn't require wires.

Most systems can either be flight packed or trailer mounted for satellite uplinking. Full time service can cost as little as $2000 per month for full-time service. For some stations that do a half dozen remotes a week, this is really not that much money. Few restriction exist as long as you can see a south sky.

Systems

Now that we have a means to get a signal where we want it let's explore a few systems in place around the country. Different formats, remote philosophies, and budgets require different systems. Some require sophisticated communications systems and cueing capability. Some are used once a month. Some stations can't justify the expense of high tech equipment. The stations presented are honest to goodness stations and provide a wide variety of remote needs.

Rock and Roll On-the-air Everywhere

This station has the requirement to be on the air remotely with very little notice. Visibility is an important effect for this station so they take a "billboard on wheels" to the events. A converted recreational vehicle, with pneumatic mast provides the ideal vehicle to accomplish these needs.

They use a 450 Mhz. type system with multiple receive sights around the city. (Figure 1) Receive sights are normally tied back to the studio via equalized phone lines. Sometimes a

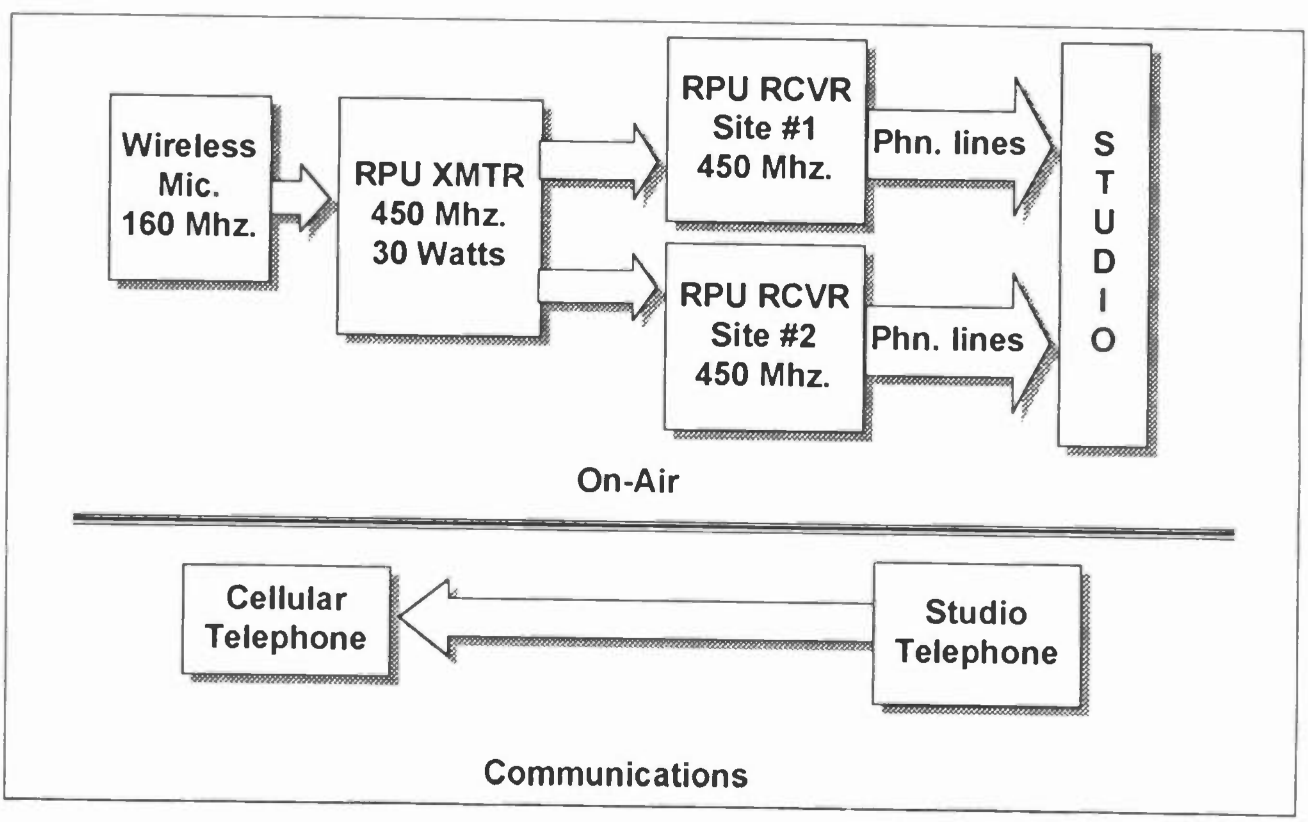

Fig. 1 - Rock and Roll On-the-Air Everywhere System

455 Mhz. link will be used in a repeat fashion. This is becoming more unusual as frequencies become more valuable.

AC Power is generated internally or the mobile unit can be plugged in at the location. Communication with the studio is provided by cellular telephone. The station is monitored off-the-air and provided to remote visitors via this off-air signal.

This is a common system used by many stations, especially music based station across the country. Proper frequency coordination is mandatory with this type of system.

A Laid Back Jazz Affair

One station we work with does about two remotes a year. Normally these remotes are in some technically civilized venue, so power and phone lines are not a problem.

We normally order in an equalized phone line, set up a mic and mixer and let them go to town. (Figure 2) This brings up the point that it is sometimes very easy for us to over-engineer an event when it's just not necessary.

This station did not even need a full-time communication circuit as their music format clock stayed the same throughout the day, so the announcer at the remote sight knew when it was time to go on the air. A cellular phone was made available for communication needed with the station.

To this station it's more important to be inside mixing with the people, than to have huge outdoor visibility. The quality audio on-the-air exists with this system, without the hassles of RF alignment or lots of wires. This can work

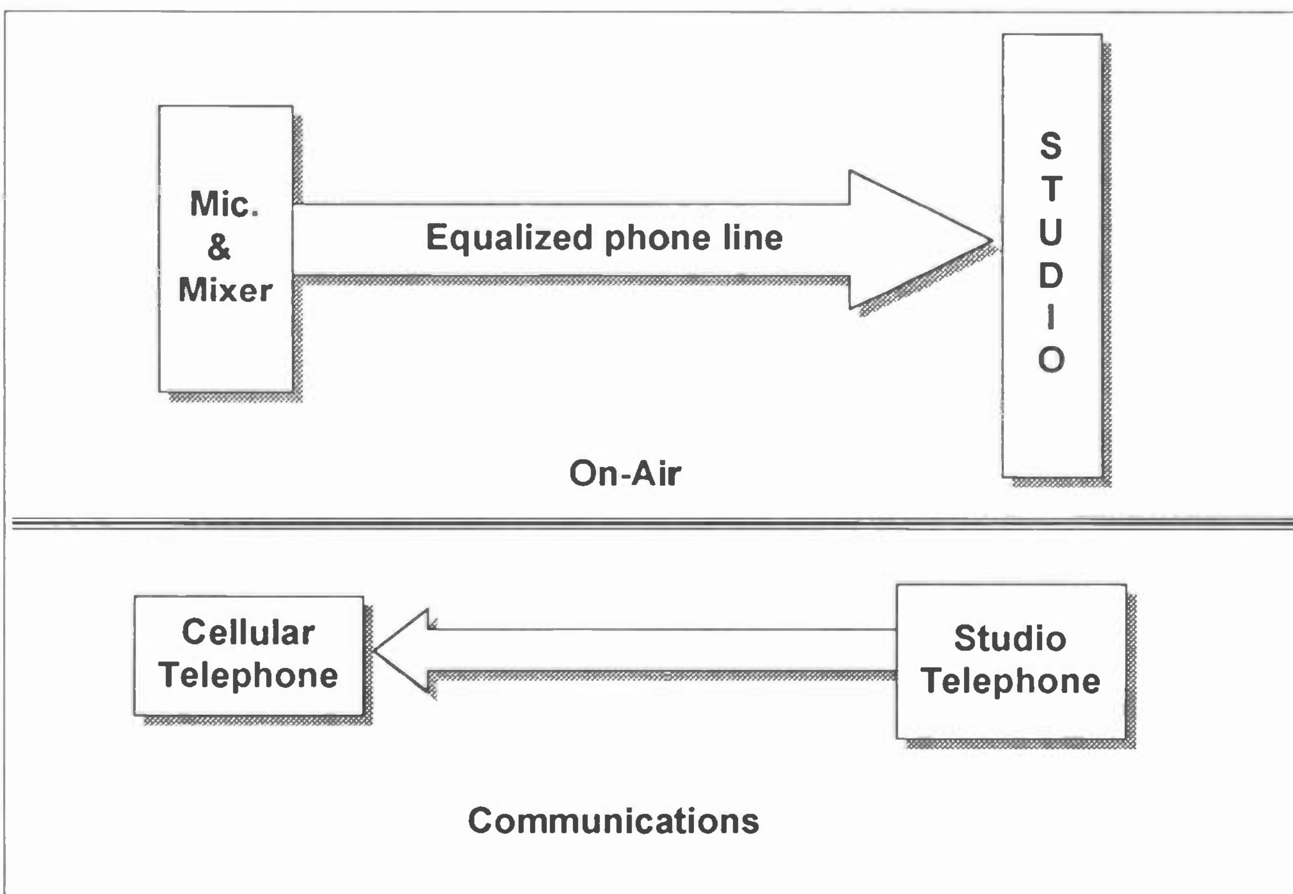

Fig. 2 - A Laid Back Jazz Affair

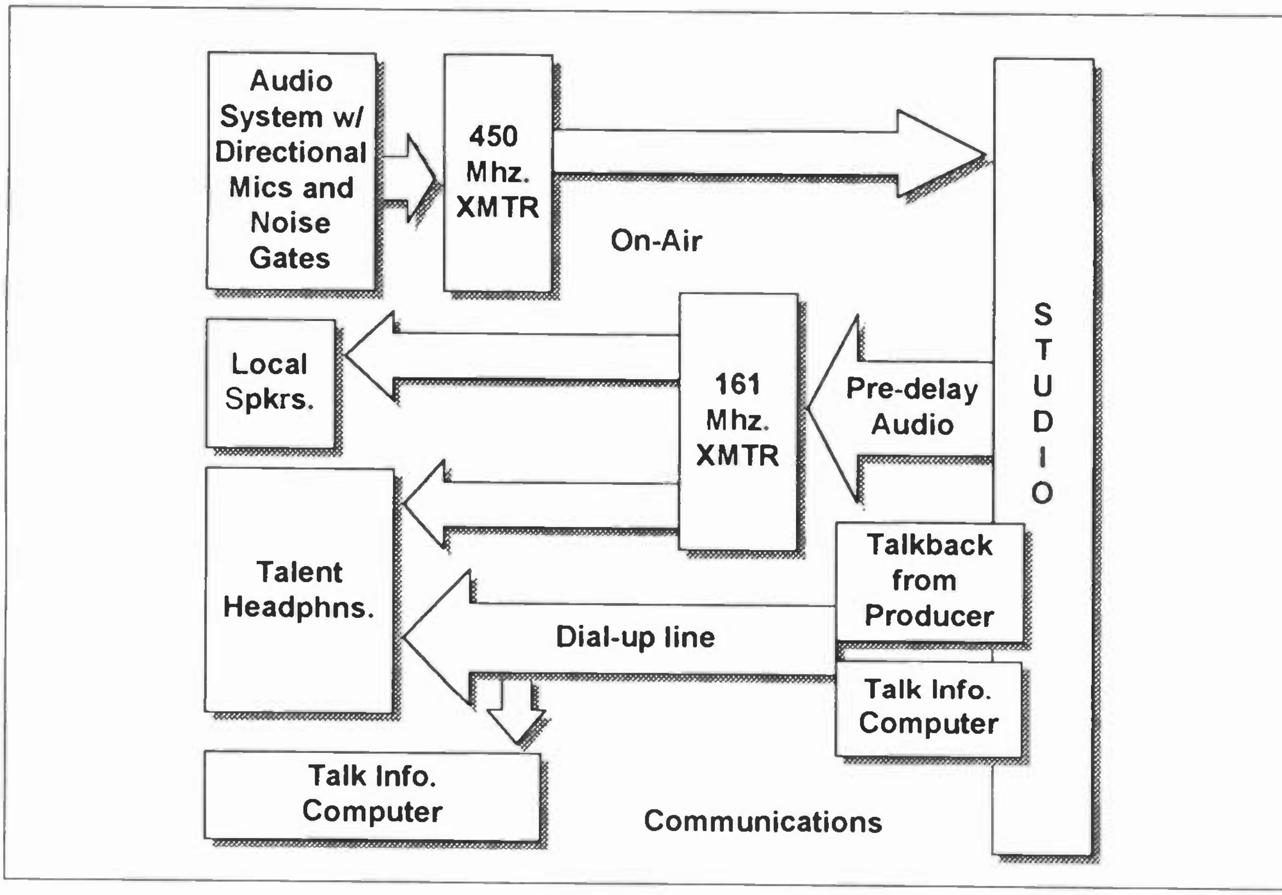

Fig. 3 - Talk of the Town

well for any station not doing a lot of remotes or for stations that broadcast from the same location on a regular basis.

The Talk of the Town

One of the more complicated formats to do "remotes" for, is the talk format. Here you are introduced to a whole other array of problems or maybe we should say challenges.

1. Most of the programming is on a profanity delay, so an on-air feed is out of the question.
2. The audio from the remote may have to be fed back to callers put on the air.
3. Hosts have to hear on-air audio, callers and cues.
4. Many times full remotes (3 hours of talk) are broadcast from locations where you can barely hear yourself think.

Normally a multiple link system is needed for remotes of this nature. (Figure 3) This can either be accomplished by radio line or multi-line dialup link.

Communication is key to a talk format. Talk hosts are used to being able to readily see on a computer screen what caller is next and have the producer cue him over his headphones. The communications system shown in the figure, provides a dialup line for both communications with the host via headphones and updating computer screens via FSK data packets sent from the studio computer to the remote computer.

Pre-delay audio is sent via a VHF, 161 Mhz. link which provides a pretty robust signal from

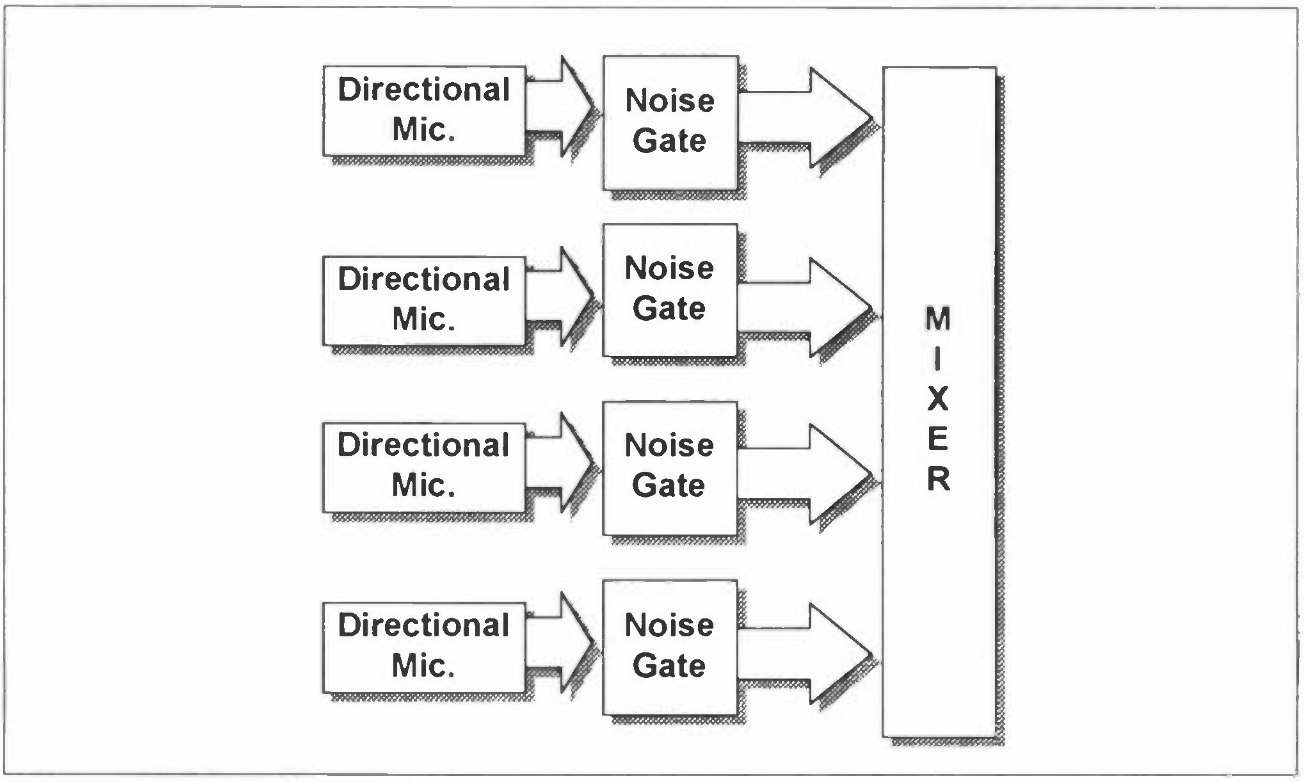

Fig. 4 - High Background Noise Environment Audio System

the studio that is more likely to penetrate buildings and make longer hauls. It's receiver is also less likely to be de-sensed by the UHF, 450 Mhz. transmitter at the remote location. This predelay audio is used for both in-house monitoring and talent headphone feeds.

A special audio rack is set up with microphone gating to help cut down on unwanted background noise. (Figure 4) Background noise can cause some awful problems back at the studio in the telephone hybrid. Seeing the background noise at the remote close to normal levels may tend to "duck" the caller or breach the hybrid isolation. Typical lower send levels to callers only make it more difficult for callers to hear talent in noisy environment, so directional microphones are handy.

Conclusions

All remote broadcasts are a challenge just due to the fact that you are walking into an unknown environment. It always helps to remember a few simple hints.

1. Always have an out. Even if this means someone at the studio taking over. This can be better than garbage on the air or dead air.
2. Try to have back-ups. Transmitters if possible, phone lines, mixers, mics, monitors, etc.
3. Allow enough time for setup. It always seems the amount of time you allow is never enough to get the job done. Rushing at the last moment before air time only makes everyone uncomfortable, especially talent.
4. Proper and adequate equipment packaging can make your life easier. Dedicated boxes for cables, mics, mixer and even RPU gear can keep you organized and help the gear to last longer.

5. Testing the location can save headaches trying to make a remote work. This is especially true for any RF shots. Even for telephone lines it makes sense to make sure all lines are in and working.

6. Communication with others staff and departments can make the difference between success and failure. Find our exact locations for talent and gear. Check on access before and after the event for engineering. Find out what type of sound system is needed for the size crowd expected.

7. Keep the gear in repair. After the remote we sometimes want to forget about remotes until we have to. Getting to that next remote and finding that the shorted monitor cable you need didn't fix itself, can be embarrassing.

Remote broadcasts can be your biggest challenge. They can also be your most noticeable success. Take the time to plan and test. Be creative in new ways to route signals. Take advantage of new low cost technologies. Keep up on new techniques and transmission schemes. Remote broadcasts are just an extension of the broadcasts from the studio so they should always sound every bit as good. And remember, keeping it bullet resistant may mean keeping it as simple as possible.

Paul Montoya is President of Broadcast Services of Colorado, Incorporated. Broadcast Services is a full service broadcast bureau providing technical services, event coordination, software development, and direct mail core listener marketing for radio and television stations. Further information can be provided by calling (303) 988-4733.

RESOLVING BROADCAST-RELATED INTERFERENCE TO CONSUMER ELECTRONICS EQUIPMENT

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Abstract

Interference from radio and TV stations to consumer electronic equipment is of growing concern to many broadcasters. As homes and offices move closer to transmitting towers, and as more new broadcast stations are licensed, electrical interference is affecting people in greater numbers. This paper reviews the thorough efforts made to address interference concerns in the neighborhoods surrounding one particular multi-user site. These efforts included attending neighborhood meetings, visiting homes to reduce interference, and evaluating susceptibility of various brands of electronic equipment to interference.

INTRODUCTION

Literally hundreds of residents can be affected by a single multi-user antenna site located in a densely-populated area. For these residents, interference is a nuisance that affects their daily lives by restricting the use and enjoyment of their electronic equipment. The interference may be relatively minor, such as a barely-detectable ghost image on certain television channels, or it may be more severe, rendering certain types of equipment such as telephones and stereo systems completely inoperable.

For broadcast stations, the ramifications of causing interference can range from receiving a few minor complaints to being involved in costly lawsuits. Interference also may result in lost revenue, as viewers and listeners tune to other stations not affected by interference. FM stations with new or modified transmitting facilities are required by §73.318 of the FCC Rules to resolve certain types of complaints received from persons located within the defined blanketing contour. Although the rules governing TV station authorizations are not as explicit (see §73.685), under some circumstances TV stations also are required to perform corrective measures to remedy interference caused by the stations. In any case, the ability to reduce or eliminate the effects of electromagnetic interference at residences or businesses is essential in order to reach a satisfactory resolution to an interference problem.

This paper discusses each of the steps used to relieve interference in an area that had been plagued by radiofrequency interference from a nearby multi-user antenna site. The procedures used to identify and reduce interference to home audio, video, and telephone equipment are described, based upon our experiences in visiting over 50 homes. The paper concludes with a list of equipment and supplies that are particularly useful for solving broadcast radiofrequency interference problems.

THE PROBLEM

At the particular broadcast site of concern, transmitting facilities of eleven full-service VHF and UHF television stations, and four FM stations are located at a single hilltop site. This ideal transmitting location is also an attractive residential area, providing a spectacular view for the local residents.

Claims of interference problems had been made since the development of the site many years before but, with the proliferation of new low-cost electronic devices such as memory-dial speakerphones, facsimile machines, and videocassette recorders (VCRs), radiofrequency interference had become a serious concern affecting almost every home in the area. By the time our firm became involved in the situation, tension had reached the breaking point. The company managing the site was seeking to enlarge the site to allow for more tenants, and had applied for city approval. Residents were furious, and the regular meetings of the neighborhood homeowners association provided an ideal forum for residents to unite in opposition to the proposed expansion. The residents successfully blocked approval of the city permit to expand the facilities, and they threatened to pursue further legal action to force the broadcast stations to reduce power or to relocate. It was in this climate that the site management company requested that we begin work to resolve the interference complaints.

We needed to identify, if possible, those stations most responsible for the interference. Residents had complained about one FM station in particular that was causing interference, so we set out to measure the relative field intensity of each station on the site at various locations in the neighborhood. Using a spectrum analyzer and a dipole antenna, blocks of spectrum from 50 to 800 MHz were measured. It was found that the relative strengths of the signals from each station varied considerably, depending upon location within the neighborhood. One or two stations could not be singled out as the most significant contributors in all cases. Additional investigation revealed that several different stations were usually involved in each interference case. Thus, resolving the interference complaints could not be accomplished by altering operations of just a few stations at the site, as some residents had suggested. Rather, cases of interference had to be addressed individually by eliminating the interference at each home.

SURVEYING THE PROBLEMS

The first step was to establish a working relationship with the homeowners association, which we did by contacting several of the association's board members and attending one of their membership meetings. At the meeting, residents aired several concerns about the site, but the main issue was interference caused to their electronic equipment. Solving the interference problems appeared to be the most important first step toward better relations between the neighborhood and stations at the site, so we embarked on what turned out to be a nine-month project to reduce and eliminate interference at each residence where it was reported to be a problem.

We had little idea what types of interference we were going to encounter. Rumors of singing bathroom fans and erratic garage door openers were circulated at the neighborhood meetings; it was clear that we needed to pin down exactly who was having interference and what types of interference were being observed. To assess these and other related issues, we developed an information survey that was mailed to over 500 residents in the area surrounding the transmitting site. A representative survey is shown in Figure 1. Only 108 surveys were returned, possibly indicating that the majority of those receiving the survey did not have interference that was severe enough to report. As expected, most of those residents that *did* return the survey reported having interference problems of some kind. The most commonly-reported interference problems involved television sets, audio equipment, telephones, and VCRs. Many survey respondents indicated they wanted someone to visit their homes to help in resolving the interference, so we began preparations.

VISITING HOMES

Before visiting any homes, we needed to define a policy that would govern what types of equipment we could and could not repair at a residence. We were concerned about the potential liability involved with entering a home and adjusting or altering a resident's electrical equipment. For example, we might need to remove or replace certain cables connecting a stereo system, but we did not want to modify the stereo system itself. Such modifications could lead to repercussions months or years later–any time the stereo malfunctioned in the future, it could be blamed on the engineer who performed the modification. We decided, therefore, to limit our actions to those that could be accomplished without modifying anything that was not intended for adjustment by consumers. To advise residents of our capabilities and limitations, we composed a standard letter that was given to each resident before we performed work. In the letter, we made it clear that no action would be taken without the resident's permission, and that no equipment would be modified. Residents were also advised that, because of the difficult nature of some problems, more than one visit to their home may be necessary before the problem could be fixed. This would allow for situations where we did not have with us the proper equipment to fix a particular problem, or for instances when we needed to construct custom solution devices in our electronics shop.

Most residents we visited were pleased to have some help resolving the interference and were more than happy to allow us to tinker with their telephones, stereo equipment, and television sets. There were some residents, however, who were unhappy with the entire situation and were displeased with spending their own time working with us on interference problems. In either case, one of the most important steps was to listen carefully to the residents' complaints. This allowed the residents to express whatever problems and concerns they had about the interference and allowed us to gauge the residents' attitude toward the problem(s). Attentive listening also helped in diagnosing the interference problems; in some cases, we were able to diagnose the causes of problems before even looking at the equipment involved.

When working on a resident's electronic equipment, we were always careful to involve that person in the process, advising them of each step. This ensured that the resident would be fully aware of what we had done with their equipment, so that they could see that we were not damaging or modifying anything. An additional benefit of involving residents in the work was that they were able to learn about the problem-solving process, so that

1. Are you experiencing problems with interference to electronic equipment at your residence?
☐ Yes ☐ No

2. Are the problems most prevalent with your:
☐ telephone ☐ television
☐ radio ☐ CD/tape player
☐ VCR ☐ other (please specify: ____________________)

3. How would you rate the severity of the interference?

barely noticeable ☐ 1 ☐ 2 ☐ 3 ☐ 4 ☐ 5 severe

4. What type of interference are you receiving?
televisions and VCRs:
☐ ghosts or double images ☐ vertical bars ☐ noise on speakers
☐ other: ____________________
phone and audio equipment:
☐ buzzing ☐ voices or music ☐ other: ____________

5. Do you have cable television?
☐ No ☐ Yes

6. Have you discovered any particular equipment brands that are...
more susceptible to interference?
☐ No ☐ Yes (please specify: ____________________)

more resistant to interference?
☐ No ☐ Yes (please specify: ____________________)

7. Have you found any effective methods for reducing the interference?
☐ No ☐ Yes (please specify: ____________________)

8. Would you be interested in attending a workshop on how you can reduce interference in your own home: ☐ Yes ☐ No

9. Would you be interested in having an engineer from Hammett & Edison visit your home to assess the nature of specific interference complaints: ☐ Yes ☐ No

Figure 1. Example of information survey sent to residents near the transmitting site.

they could solve some types of problems on their own. Educating residents about the causes of interference also took the "mystery" out of the problems that they had been experiencing, alleviating some of their concerns about the interference. Before leaving a home, we were careful to demonstrate to the resident that each piece of equipment we had worked on was still functioning normally (or the way it had been before we entered the residence). This reduced the potential for later claims that we had damaged their equipment.

ELECTRONIC EQUIPMENT INTERFERENCE

We visited over 50 homes and were successful in fixing the majority of those problems that were interference-related. Figure 2 shows the relative percentages of problems relating to each type of equipment. Although television problems were the most frequently reported on the information surveys, our visits revealed that the majority of the problems–over 40% of all observed problems–related to telephone equipment (telephones, facsimile machines, cordless telephones, etc.). Television sets were in second place (23%), followed by stereo systems (12%), VCRs (4%) and other types of equipment. Surprisingly, 18% of the problems were not related at all to interference. The following sections provide details on our progress in finding ways to resolve these problems and to overcome the susceptibility of various types of equipment to VHF and UHF interference.

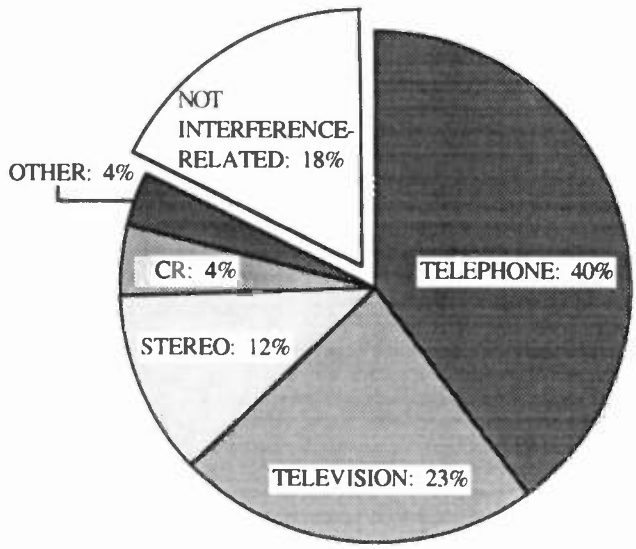

Figure 2. Types of interference problems observed.

Telephones

The addition of active electronics in telephone equipment has brought numerous features to telephone sets, but the presence of diodes and other semiconducting devices also has brought greater susceptibility to radiofrequency (RF) interference. When electronic telephone sets are exposed to broadcast RF fields, the semiconducting devices in the telephones can rectify the fields and feed demodulated audio to the telephone handset, where it is heard as background noise. Electronic telephones can detect both AM and FM modulation, so the resulting interference from nearby VHF and UHF stations can sound like a mix of music and voices from FM stations combined with buzzing from television sync pulses. Almost every home we visited had some sort of telephone interference and, since many residents near the site had businesses in their homes, these telephones were also important occupational tools. Solving telephone interference problems, therefore, was one of our most important goals. Fortunately, eliminating interference from telephones was usually a straightforward process, the success of which depended mostly upon the type of telephone involved. Older rotary-dial and DTMF telephones such as the Western Electric 500 and 2500 sets received almost no VHF and UHF interference, while some newer telephones were rendered completely useless by strong interference. Some telephones could even be tuned to receive specific stations by adjusting the length and orientation of the handset cord! Most telephones were found to be somewhere between the two extremes and could be improved using proper techniques.

Since we were limited to using methods that would not require modifying the internal circuitry of the telephones, the goal was to keep the RF energy from entering the telephones at all. It was found that the RF energy usually entered the telephones on the handset, line, or power cords, each of which could act as an antenna. To determine which of the cords carried the interference, we varied the length and location of each cord, paying careful attention to how (or if) the interference changed. If holding the handset cord with one hand changed the degree of interference detected, then RF energy was being carried by the cord. Likewise, moving the telephone line and power cords around indicated if they were involved in the problem. Once the offending cord was located, wrapping the cord around a snap-together ferrite filter usually reduced or eliminated the detected interference. Sometimes it took a bit of experimenting to find the optimum location and number of filters, but a single filter located as close as possible to the telephone set usually worked best.

Ferrite filters act to block the flow of RF current in wires because they appear as an inductive reactance in series with the wire. At radio frequencies, this inductance can present an impedance of 100 ohms or more to the RF current. Wrapping a wire through a filter several times increases the impedance, so this is a good technique for getting the most suppression from a single filter. Multiple filters can be placed on a particular wire for even greater suppression. Ferrites are available in a wide variety of shapes and sizes (see Figure 3), and different materials are effective over various frequency ranges. We experimented with ferrites from several manufacturers, and found that the Radio Shack snap-together types worked as well as any. One important advantage of the Radio Shack filters is that they are readily available. Therefore, they could be recommended to those residents who were interested in resolving interference on their own. The availability of the filters also made the interference-reduction process seem less complicated, since we were not installing some mysterious device, but something that anyone could buy at a local store.

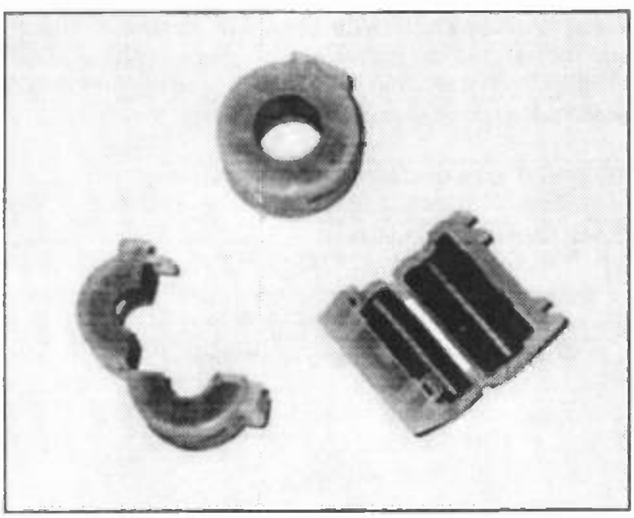

Figure 3. Typical snap-together ferrite filters.

Other telephone-related devices such as answering machines, facsimile machines, modems, and cordless telephones can suffer from some degree of RF interference. For all but cordless telephones, the techniques described above for standard telephones were found to be effective. For 49 MHz band cordless telephones, there were fewer effective non-invasive methods for reducing interference, since they are by nature more sensitive to RF fields. We believe that some of the telephones suffered from overload in their RF front-end circuitry because of the presence of a strong Channel 2 signal (54 to 60 MHz). Other cordless telephones had the same problems as corded telephones in cases where RF was being conducted along the line cord, and ferrite filters helped in these instances. We found that most cordless telephone interference was received by the RF circuitry, however, and could not be reduced by using ferrite filters. Relocating the telephone base unit to another part of the room was effective in some cases but, typically, the only solution was replacement of the telephone with another, more resistant model.

For cordless telephones, or in those cases where filters do not reduce interference on standard (corded) telephones, there are certain brands and types of telephones that can be recommended as replacements. For standard telephones, the safest recommendation is a Western Electric 2500 set, or the equivalent ITT model available through telephone equipment distributors. For those who cannot live without speed dial, hold, and a speakerphone, the Radio Shack Duofone-148 worked much better in high RF fields than equivalent or more expensive models from other manufacturers. For 49 MHz cordless telephones, certain brands such as the AT&T 5400-type were found to be more resistant to interference and performed well in areas where other brands were unusable. Another possible solution–one that we have not yet investigated–would be to use a 900 MHz cordless telephone, which would probably be less sensitive to VHF-band interference.

Audio Equipment

Interference to stereo systems is similar in some respects to telephone interference, because the audio sections of the stereo circuitry are usually affected. As with telephones, the interference often appears as buzzing or background audio on the speakers. This can be particularly disturbing when listening to classical and other musical formats characterized by low-level audio or frequent periods of silence. To reduce the interference, one must prevent the RF from getting to the audio stages of the equipment. However, because there can be several interconnected components in a stereo system, each of which can contribute to the overall problem, stereo system interference problems are usually more difficult to cure than telephone problems. Random placement of ferrite filters can be unproductive and frustrating; a methodical, logical approach to the problem is more effective. We found that the best strategy was to simplify the installation as much as possible by temporarily removing equipment from the system in order to find where the interference was entering. Once the problem was isolated, ferrite filters or other means could be used to alleviate the problem.

A typical stereo system consisted of a tuner/amplifier, a CD player, a phonograph, and two speakers. The first step was to begin disconnecting the components, listening for any changes in interference. If the interference lessened or disappeared when the CD player was disconnected, for example, it indicated that either the CD player or the patch cables connecting the CD player to the stereo were carrying the interference. We would continue disconnecting components until the interference disappeared completely, or until we were out of components to disconnect. If interference was present with just the amplifier and speakers, this meant that the RF energy was conducted along the power cord or speaker wires or was radiated into the amplifier directly. In almost all such cases we found that the RF energy was conducted along the speaker wires, detected in the amplifier stage, and sent back out to the speakers as audio interference. Ferrite filters placed on the speaker wires close to the amplifier helped reduce the level of interference in these cases. For some stereo systems, the interference was much worse on one speaker than the other, and sometimes merely swapping the connections (hooking the right and left speakers to the left and right speaker terminals on the back of the stereo) reduced the interference. Another trick that worked on some stereo systems with A/B speaker systems was to connect the speakers to the alternate, or "B" speaker terminals.

Once the bare-bones system was free from interference, the components were added back to the system, one at a time. If interference returned, we would experiment further with that piece of equipment, changing the way it was connected, or varying the locations of the interconnecting cables. Sometimes merely tidying up the wiring by bundling excess cable length with cable ties helped enormously. Listening to the relationship between volume control and interference level provided valuable information about the location of the interference reception, also. If the volume control had no effect, it indicated that the interference was being received by the final amplifier stage and that there may be a ground loop problem or speaker wire pickup. If the volume control changed the level of the interference, it meant that the interference was being received earlier in the audio chain, possibly in one of the other components. For example, RF energy could be received by the CD player, rectified, and sent to the amplifier as audio noise, in which case a filter somewhere on the CD player patch cords or power cord could reduce the interference.

Radio tuners also were affected by strong RF fields, although the problem appeared to be more common with small portable radio/tape players than with table-top models. Some portable radios we observed had interference only when they were connected to an AC outlet, and ferrite filters on the line cord helped this type of problem. Others experienced front end overload and would receive no FM stations at all when located in high field areas. On some portable stereo cassette players, buzzing could be heard through the headphones. Radio Shack filters helped this problem but had the disadvantage of being inconveniently large in some cases. Smaller ferrites would probably work as well, although we did not have the opportunity to experiment with this option.

Television Sets

Television interference was most prevalent among cable television subscribers, and it usually appeared as ghost images or vertical bars displayed on the pictures of stations that were both broadcast from the nearby site and sent over the cable system on the same channel. Ghosts were created because the TV set was receiving two strong signals: one directly from the site and one (delayed) signal from the cable system. Even if no external antenna was connected to the TV set, the signal from the site was strong enough to be received directly by the tuner, causing ghosting or distortions in the cable picture.

We found at least two effective methods for eliminating this type of interference. One method involved disconnecting the cable signal from the TV and using an external antenna when stations broadcast from the site were viewed. This eliminated the possibility that the cable signal would interfere with the direct signal. Disconnecting the cable could be accomplished with a video "A/B" switch, with the cable connected to "A" and the antenna connected to "B". Remote control A/B switches are available to those residents wanting the added convenience.

The other effective method for removing cable ghosts required a VCR or separate "cable converter" unit. The cable television signal was connected to the VCR or cable converter, and the output of the VCR or cable converter was connected to the television set. Tuning the TV to Channel 3 and using the VCR or cable converter as a tuner avoided the problems associated with the poorly-shielded TV tuner. In most cases, VCR and cable converter tuners were found to be better shielded than those in TVs, in that they did not pick up signals efficiently without an antenna or cable connected.

Yet another "solution" to ghosts was found by a resident who reported that his TV reception improved on some channels when his video game was turned on. After investigating his claims, it was found that by turning on the video game, the cable signal was effectively disconnected from the TV. Even with no cable or external antenna connected to the set, he was able to receive good quality pictures on those stations broadcast from the nearby site. This configuration can be achieved with a VCR or cable converter, also. Turning on the VCR or cable converter, but still using the TV tuner to select the channel, has the same effect.

Sometimes cable television ghosts were caused by signal leakage into the cable itself, which is known as cable ingress. In such cases, the first step was to investigate the location of the leakage. If the leakage occurred in cable-company-owned and -installed cable, then it was determined to be its responsibility to prevent the ingress. In the neighborhoods we visited, the cable company was familiar with ingress problems and was well-equipped and willing to solve such problems. Alternatively, if the cable ingress occurred on poorly-shielded cables connecting components such as VCRs and TVs, we would recommend that the resident install a better type of cable. We found that Belden 9054 Duobond Plus RG-59-type cable worked very well for this purpose. Another source of ingress was open taps on wall outlets or on splitters. Shielding all unused cable outlets or splitter taps with 75 ohm terminations helped reduce ingress in some cases.

Television audio circuits sometimes suffered from the same problems as stereo systems, receiving FM stations or loud buzzing that could be heard through the TV speakers. We could do little about this problem without opening the television sets, so we were limited to making recommendations about actions that could be taken by authorized personnel. Some technically-capable residents implemented our suggestions and had success with ferrite filters placed on the speaker wires inside their television sets.

VCRs and Camcorders

The problems experienced with VCRs and camcorders were among the most difficult to deal with because external filtering usually did not reduce the interference. For camcorders, no external filtering was possible since camcorders are typically compact, self-contained units. Fortunately, camcorder interference was uncommon in the neighborhoods we visited; only one resident reported having problems, and he discovered another brand that did not receive interference. Certain brands of VCRs, too, were found to be less sensitive to interference, and in most cases of VCR interference the only viable solution was to purchase a more interference-resistant unit. VCR interference was more common than camcorder interference, though, and we had success in solving some of these problems.

In cases where external wiring brought unwanted RF signals into VCRs, reconfiguring the way the VCR was connected to the television set (or to other VCRs) seemed to help. In one particular case, audio interference appeared on programs recorded from one VCR to another, but routing the signal from the playing VCR through the television set, and then to the recording VCR, cured the problem. However, most cases of VCR interference that we observed were due to direct pickup of interference by the VCR's internal circuitry. The symptoms of this type of interference were poor playback video quality, video with white "sparklies", and video tapes that would not play because the VCR could not lock onto the proper tape speed. In this case, the VCR would flip between SP, LP, and EP modes, never settling on one speed. Some of these problems could be fixed by relocating the VCR within the room or to another location in the house. We experimented with our own VCR, which suffered from the same tape speed problem, and found that it could be remedied by enclosing the entire VCR in an aluminum box and by grounding the aluminum box to the 75 ohm F connector on the back of the VCR. While this was a solution, it was not something that we could implement easily at residences, since the shielding was both mechanically and aesthetically undesirable. A better solution was to purchase another brand of VCR that was more resistant to interference. For example, we found that the Mitsubishi model U34 VCR worked well in those locations where other brands of VCRs had failed.

Other Reported Interference

Several other types of electronic equipment received interference from high VHF and UHF fields. One such problem was with wired intercoms, which suffered from buzzing and radio background noise. At the locations we visited, the problem was particularly difficult because there were many interconnected devices throughout the house, each of which contributed to the overall problem. We were able to improve the audio quality in these cases by using ferrite filters on the wiring, applying a methodology similar to that used with stereo systems.

Interference was reported on other devices, such as car security systems, electronic keyboards, garage door openers, and almost anything else possessing an AC line cord or batteries. Some of these reports turned out to be genuine interference problems. After visiting several homes, however, it became apparent that there was often a difference between *perceived* interference-related problems and *actual* interference-related problems. For instance, a telephone may have been malfunctioning because of a worn handset cord, not because of strong signals from stations at the nearby transmitting site. Many problems were indeed interference-related, but quite a few were caused by "user error". We received several complaints about malfunctioning garage door openers, and in every case the problems were not related to electrical interference at all. The actual problems included missing or broken receive antennas, dead batteries in remote controls, and mechanical failures in the garage door mechanisms or remote controls. Other residents complained about malfunctioning infrared remote controls; the causes of the malfunctions were found to be similar, involving old batteries or simple mechanical failures.

Residents' misconceptions of interference were often fostered by electronics salespeople and repair technicians.

Some residents had complained to the stores where they purchased their equipment and were told by the salespeople that no equipment would work perfectly near the transmitting site. Others were told that their electrical circuitry would wear out faster near the site. We were often in the position of educating residents not only on how to solve their interference problems, but on what interference problems actually were. This was difficult at times, since to some residents it may have seemed that we were "covering up" for problems caused by stations at the site. Even in those cases, however, an honest, open approach usually got us through the uncomfortable situation.

Poor television and radio reception were among the most commonly reported problems not related to interference. Television reception near the transmitting site was perfect at some homes, while terrible at others. The main problem was severe ghosting, due to strong reflections from surrounding buildings and terrain. Using directional receiving antennas helped, but the best antenna orientation differed for each channel. While this was not really an interference problem in the traditional sense, it was an aggravation that residents near the transmitting site had to cope with. Cable television reception was typically much better in those cases where the cable ghosts could be eliminated using the methods described above.

Radio reception in the area seemed to be better than TV reception, and most of those problems we observed resulted from disconnected or missing antennas. One interesting fact was that interference to all equipment was less severe on the lower floors of a house. Moving equipment from the upper level to the basement sometimes made the difference between a completely unusable piece of equipment and one that functioned perfectly. Since the homes were often located on hillsides, the lower floors were partially surrounded by earth, which provided additional shielding of the signals broadcast from the site. Although residents were usually not willing to move their living rooms to the lower floors, sometimes interference-susceptible telephones could be swapped with less fashionable but more functional telephones that formerly had been relegated to the basement.

USEFUL EQUIPMENT

During the several-month period when we were visiting homes, we compiled a list of the equipment found to be most useful in dealing with RF interference problems. A listing of some useful equipment we brought with us during the visits is shown in Table 1.

The ability to test electronic equipment in the field was important when we needed to evaluate a particular model or brand of equipment in high RF environments. A

Table 1. Useful interference-reduction supplies.

Telephone Supplies
Western Electric or ITT 2500 set
Radio Shack Duofone 148 telephone
AT&T 5400 cordless telephone
Modular plugs, handsets, and line cords

TV/Radio Supplies
UHF/VHF 300 ohm combiner
75/300 ohm balun
300 ohm twinlead
300 ohm folded dipole VHF antenna
75 ohm Belden "Duobond" 9054 cable
3 dB 75 ohm splitter
UHF/VHF "rabbit ear" antenna
75 ohm A/B switch

TV/VCR Supplies
VCR, Mitsubishi Model U34
TV monitor
VHS test tapes

Filters
Ferrite filters, Radio Shack #273-104/105
Ferrite filters, FerriShield material #28
Ferrite filters, Fair-Rite #0443164151
AC line filter, Radio Shack #15-1111

Useful Equipment
AC power inverter
Spectrum analyzer
Field strength meters
Digital multimeter

Tools
Philips and slotted screwdrivers
75-ohm cable wrench
Cable cutters/wire strippers
Soldering iron

Miscellaneous Supplies
9V batteries
Infrared sensor, Radio Shack #276-099
Duct tape
3M Copper tape
Electrical tape
Cable ties
Aluminum foil
Power line tester
Miscellaneous audio and video cables and connectors

power inverter connected to our 12 volt automobile battery allowed us to test television sets, VCRs, and other AC-powered equipment in our field truck. Two nine-volt batteries connected in series provided ample power to run telephones, and we discovered that an interference-susceptible telephone (AT&T Model 710 in this case) powered by batteries, made a relatively inexpensive and useful indicator of RF field levels. We could use the test telephone to locate residential areas likely to have interference problems, or we could demonstrate to residents the types of interference possible on electronic telephones.

A handy device for diagnosing infrared remote control problems is the infrared sensor card available from Radio Shack. This card has a small patch of material that glows when illuminated by infrared radiation, and it makes checking a questionable remote control a simple task.

Other useful equipment included a good assortment of tools, cables, and connectors. In most cases, however, we used just a fraction of the equipment we brought with us; a couple of screwdrivers, several ferrite filters, and some cable ties solved the majority of the problems.

RESULTS

After compiling the results of the information surveys and visiting several homes, we attended a second homeowners association meeting and made a formal presentation on the results of the surveys and visits. A major part of the presentation was a tutorial session on methods for reducing interference in homes. The meeting was well attended and the residents seemed very receptive to the presentation. It helped, of course, to have visited the homes of the association's board members and to have resolved *their* interference problems. Those board members present at the meeting spoke well of our abilities, and many more residents requested that we visit their homes based upon our success in the initial cases. Following suggestions we made during the meeting, some residents were able to remedy interference problems on their own by properly installing ferrite filters on their telephones and audio equipment.

Upon completion of our visits to homes in the area, almost all interference problems had been resolved; for those remaining problems, we had made specific recommendations for further action, such as contacting the local cable company or replacing susceptible equipment. In the end, most residents were satisfied, and the topics at the monthly homeowners association meeting shifted from interference to other issues not related to the nearby transmitting site.

In the several months since our last visits to homes, we have received occasional calls from residents wanting our advice on what types of equipment to purchase. Only one resident complained that his stereo system did not function properly after we worked on it. Much to the resident's embarrassment, a quick inspection of the unit revealed that one of the speaker wires had come unplugged sometime after our visit.

CONCLUSION

Any radio or television station would be wise to investigate reasonable claims of interference received from residents or businesses near the station's transmitting site. The obvious benefit of such a policy would be to prevent problems from reaching the critical stage where legal action is involved. Preventing a lynch mob from forming in the first place will save time, effort, and money in the long run.

Some useful information on interference problems can be found in FCC and ARRL publications [1, 2] and in manufacturers' association literature [3]. While this information is geared more towards low frequency (MF and HF) interference, and does not provide much detail on VHF and UHF interference, many of the same principles apply. Solving the problems caused by RF interference can be a trying process but, with logic and patience, a competent engineer almost always can find a solution.

REFERENCES

[1] *Radio Frequency Interference: How to find it and fix it*, American Radio Relay League, Newington, CT, 1991.

[2] *Interference Handbook*, Federal Communications Commission, Washington, D.C., 1986.

[3] *Consumers Should Know Something About Interference*, Electronic Industries Association/ Consumer Electronics Group, Washington, D.C., 1987.

FREQUENCY COORDINATION: THE BEST SOLUTION TO INTERFERENCE

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Abstract Modern Part 74 coordination began in 1976. Painfully at times, volunteer coordination entities have devised tactics to maintain relative spectral peace and harmony. Their foremost challenge: To accommodate a growing array of users in a finite amount of spectrum. Resolving interference peacefully has been a necessary goal within that challenge. A basic level of trust between licensees, like critical mass in a nuclear reaction, is needed to hold the process together. Without peaceful resolution of problems that arise between licensees, the process falls apart. This paper will outline key concepts, principles, and strategies designed to meet this challenge and accomplish the goal. The case study method will be used to demonstrate practical applications of some of these tactics.

INTRODUCTION

What Makes Part 74 Different?

This section of the FCC's Rules is different from sections that deal with two-way or common carrier operations for non broadcast licensees. For instance, unlike other FCC regulated services, there are no exclusive channels in Part 74, only protection for exclusive paths for fixed links.

Coordination Is A Do It Ourself Project

The FCC's attitude toward Part 74 frequency coordination and interference control resembles its oversight in the Amateur Radio bands. As is the case in Amateur Radio spectrum, the FCC has very little involvement with day to day coordination and interference resolution. Ever tightening budget constraints, limited by a number of non-broadcast related activities, are some reasons FCC Field offices do not have the personnel, proper equipment, or mandate from Washington to resolve most Part 74 interference cases. Coordination and interference control is really left to local broadcasters to work out. [1]

Part 74 Licensing Is Unique

The Part 74 licensing process has not been without its problems. There have been documented instances when the FCC has issued licenses based on defective Form 313 Part 74 applications. Information on such defective applications may show a direct conflict with an existing Studio to Transmitter (STL) path that would be apparent only to engineers familiar with local conditions. While there is some evidence that the FCC has recently begun to make changes to its Part 74 licensing policies, the real burden of making Part 74 coordination work still rests on dedicated unpaid volunteers at the local level. Further education is needed so more licensees will be aware that a local resource may exist to make it easier to design a successful system.

The Need For Local Coordination

The need for Part 74 frequency coordination existed from the instant two radio stations found they were attempting to use the same VHF Remote Pickup (RPU) channel at the same time. Although history is cloudy on when and where that happened, it likely took place in Southern California. Perhaps this helps explain why broadcasters in this region were eager to form what became the Southern California Frequency Coordinating Committee (SCFCC) when the FCC's Report and Order to "split" channels in the 450 RPU band took effect in 1976. [2]

How Were The Basics Established?

Early in its history, the SCFCC established the principles of local Part 74 Coordination. Their premises and practices have become the bedrock supporting Part 74 coordination across the United States. This paper will review these basic principles in detail to understand why Part 74 Coordination volunteers comprehend the eternal truth that "No Good Deed Goes Unpunished".

Ask Not What The FCC Can Do For You...

The Federal Communications Commission looks to local coordination groups to be the court of first jurisdiction for Part 74 conflict resolution. [3] It is in the best interests of all broadcasters to have conflicts resolved at the local level. Once escalated for resolution, FCC solutions imposed from above and outside may be bitter medicine for all parties. [4]

THE FOUNDATION

Cementing The Contact

The Basic premise of local coordination is *Licensee To Licensee Contact*. Local coordination entities never actually assign frequencies. If Part 74 were constructed otherwise, they would be called *Frequency Assignment Committees!* The primary purpose of local coordination bodies is to establish and maintain an accurate local database. Licensees themselves must contact the co channel and adjacent channel licensees who will be affected by a proposed new use. This basic responsibility should be discharged soon after the Licensee wishing to make a change gets enough information on pertinent co and adjacent channel licensees from the local database. [5]

Once the idea of Licensee-To-Licensee Communications has been indelibly etched into the minds of all licensees, the local body should offer an assortment of detailed principles to make that premise work at the emotional level.

Principles of Local Coordination

The Local Priority Principle
This principle protects local licensees when interests of Network or out of area broadcasters might clash. For instance, local coordination entities are often helpful to networks doing sports remotes. They can let networks know which local stations might not be using Part 74 channels at the time of the event. Since many sporting events occur on evenings and weekends, accommodations can be reached in most cases. [6] Once this information is obtained from the local coordination entity, the network must observe the rules of *Licensee To Licensee Contact* by contacting all parties that might be affected well before the event takes place. Many local groups also work "real time" during such events to address problems that inevitable arise before they escalate into armed conflict.

The First In Priority Principle
A licensee with a prior existing use in general has priority over another licensee requesting a new use.

The Usage Priority Principle
A hierarchy of uses is outlined under Section 74.403 (b) of the Rules. Emergency communications have top priority. Tests or drills to check the performance of stand-by or emergency links have the lowest priority. The FCC has used and will use their priority list to resolve conflicts that cannot be resolved at the local level. [7]

The Protection Principle
A licensee with a new intended use may offer to upgrade equipment for existing licensees who may be affected by the proposed new use. The existing licensees may not unreasonably withhold testing and implementation, if such protection is offered. However, licensees proposing a new or modified use must not be coerced by existing licensees to provide such equipment upgrades.

The Good Engineering Principle
Licensees who have a clean house from an engineering practices point of view generally receive more support from local coordination bodies than those who do not.

The "Elmer" Principle
Like proverbial old-hand "Elmers" who help new amateur radio operators, we must educate broadcast engineers who need training to deal with modern Part 74 operations, especially in regions experiencing the most growing pains.

The Clearinghouse Principle
Local bodies should act as clearinghouses for the coordination process. New and existing licensees should keep the local body informed of negotiations in progress. Written confirmation is encouraged from all parties as the process unfolds.

The Town Crier Principle
When changes to the database occur, or are proposed, the entire licensee membership of the coordination entity should be notified. Many local entities publish monthly newsletters for this purpose.

The Open Arms Principle
Who are the members of local coordination bodies? Every Part 73 licensee, qualified network, CARS band authorized cable entity, and other party who derives licensing permission through Part 74 are members. The legal principle is one of inclusion, rather than exclusion. Any hint of favoritism or restraint of trade must be avoided in membership rules. For further information, seek qualified legal assistance.

The Oversight Principle
The local coordination entity should be notified in advance by licensees when they file FCC Form 313 for a new or modified use. Copies of Form 313 should be filed with the local body in advance of filing with the Commission. If a Form 313 has been filed with the FCC and the local body finds it has not been notified, it should review the situation with the licensee and the Commission to assure answers to informational Question 16 on the 313 Form agree with information on file with the local coordination body.

The Caesar's Wife Principle
Like the legendary wife of the Roman Emperor Caesar, the local body should represent a high moral standard. Its words and deeds must be above reproach when inevitable conflicts arise between licensees. It must be an effective and respected

mediator. Since local bodies consist of other broadcasters, this is not as easy as it sounds. Some groups use the Society of Broadcast Engineers Canons of Ethics as a bench mark to measure their immunity to real or potential conflicts of interest and their conduct towards fellow broadcasters. [8]

The Dutch Uncle Principle

While local bodies should never be perceived as "Frequency Police", it is inevitable that Part 74 Rules violations or potential violations will be brought to their attention. Local bodies should not be silent or willing parties to operations that are at odds with Part 74 Rules. When this happens, the entire range of potential actions and options should be carefully evaluated before anything is done.

DIPLOMACY AND LOCAL COORDINATION

No Diplomatic Immunity In Part 74

The art of diplomacy is essential when conflicts come to the attention of the local entity. Diplomacy is merely being sensitive to what is said and how it said, before it is said. While it is beyond the scope of this paper to give specific advice on legal issues, some special coordination corollaries to the basic rules of diplomacy should be observed:

Newton's Laws

Every statement has the potential to create an equal and often opposite, unpleasant reaction to what is necessary to resolve conflict. Gravity rules. What goes up, must come down. Familiarity with local terrain is needed to predict where impact will occur. Plotting the equations of conflict in advance to predict what might happen is always a good idea.

Sticks and Stones

Careful choice of words is vital once a course of action is decided on. The wrong word at the wrong time may rupture the mediation process. Since perception becomes reality, it is a good idea to tape record meetings to identify when, where, or if misinterpretation has occurred. Accurate note taking and record keeping is also important since what has transpired might find its way before the Federal Communications Commission, or become evidence in a potential law suit. All parties to the conflict should therefore submit their positions to the coordination entity as well as each other in writing.

Fairness Doctrine

At some point, a fact finding meeting should be arranged under the auspices of the local body. The time, place, and who is invited to attend are all critical factors. Any members of the local body who may be connected to the dispute should excuse themselves from presiding at such meetings. A neutral party has to preside. This person must keep the meeting on track and constantly reassure the attendees that the process will unfold without prejudice. Arbitration takes place after the fact finding portion of the meeting if events progress smoothly.

Undercover Agents

Sometimes a well placed word from a friend to a party to a dispute can help. Local coordination volunteers should educate themselves about the social side of the local engineering or news communities during conflict resolution.

Everybody Wins

Successful negotiators know that the best negotiations are the ones where everyone wins something, and no one loses too much. The secret is to find out what everyone really wants and find a way to let them have it in some form. Carrots are preferred over sticks in most negotiations. If a station wants a 450 MHz channel to contact its field crews, suggesting a trunked dispatch system might be a way out. The additional privacy from scanners that trunked systems offer might be an effective carrot.

About Face

The oriental concept of "Face" is very critical. Few people like to admit that they have been pig headed, wrong, ignorant, blind, or stupid. Solutions that reflect any or all of the above on one or all of the parties probably won't work. Mediators who are successful are usually skilled at "saving face".

If All Else Fails

The Society of Broadcast Engineers offers its Washington legal counsel as a resource to local coordination entities under certain circumstances. The amount of help possible will usually be inversely proportional to the amount of trouble the local entity has managed to get itself into by not observing the guidelines outlined above.

When All Else Has Failed

While the FCC remains as the final arbiter of spectrum conflicts, their pattern has been to turn some disputes back to local coordination bodies for another mediation attempt. Expect lengthy delays if events bring matters to their attention.

Support From Above

The SBE National Office makes the following resources available to assist the local coordination process:

- General information on the coordination process has been available in cooperation with the NAB since the 7th Edition of the their Engineering Handbook. [9]

- A national Frequency Coordinator List is regularly updated, published quarterly, and circulated to a large mailing list that includes the FCC.

☐ A compiled database program is supplied, free of charge, to bonafide local coordination bodies to make their duty of assembling and maintaining a database easier.

☐ A publication is available to outline the mechanics of how to set up a local coordination group.

PRESSURES ON PART 74 SPECTRUM

All The Traffic Will Bear

With local TV news at almost every hour of the day and more and more TV entities competing for ENG microwave spectrum in major markets, de-facto exclusive channels are becoming relics of the past. Home Channel Plans that allow for real time coordination are now being viewed as the way to make the best of a situation that seems to get worse every time a new user or use comes before a coordination body. Broadcast engineers are caught in the middle. They are told by TV news directors to provide ENG facilities, or else. Hell hath no fury like a TV News Director who can't get on 2 GHz.

Radio is not immune. Traffic reports every ten minutes, 24 hours a day, have been a fact of life in Los Angeles for several years. It is not unheard of to have several traffic services doing airborne reports in the 160 and 450 RPU bands for a dozen or more radio and TV stations.

The FCC Giveth...And May Taketh Away

All of Part 74 is vulnerable to the political and economic pressures and realities of a spectrum-hungry nation. So-called Personal Communications Services that will need wide band wireless communications will be with us soon. While we have been successful so far in making the case that Part 74 spectrum is the "back stage" supporting spectrum for Part 73, we should expect a steady stream of demands for what little spectrum we have, and a parallel continuing review by the FCC of our progress in using it efficiently. We may find some of us, and some of our companies, being a part of that pressure as technology marches forward. This means we must vigorously look for new ways to implement Part 74 using some of that same new technology. Compression techniques and spread spectrum may indeed let us do the impossible: use the same spectrum in the same region.

Power Corrupts

Part 74 allows up to 15 watts transmitter output power (TPO) for airborne platforms, regardless of operating altitude. [10] Since line of sight conditions exist between many major markets, it is easy to see (literally) why there are not enough channels for everyone to have their own. Serious conflicts can and do exist between stations needing channels for traffic reports and stations requiring channels for news and more traditional program length remote pickup broadcasts. The power issue also arises for mobile and repeater operations. Some licensees still use high transmitter power to overcome receiver design, installation or location problems. Section 74.461 (b) states in part "... The authorized transmitter power for a remote pickup broadcast shall be limited to that necessary for satisfactory service...". While the top power limit is 100 watts, this is overkill, even for most mountain top repeaters and mobile vehicle installations. Many users have had good luck with "receive only" sites where desensitization effects of nearby transmitters do not exist because there are no nearby transmitters. There are no easy answers when these matters come before a local coordination entity.

RELATIONS WITH THE FCC

Mission Impossible

The FCC relies on the local voluntary Part 74 coordination process to fill in Part 74's gaps on interference and coordination. While local coordination is mentioned in the FCC's Rules, no real authority is granted to local groups. [11] Their power lies in a good track record and the trust such a track record can build. Successful major event coordination such as the 1984 Olympics, numerous Space Shuttle landings at Edwards Air Force Base, and many marathons, parades, and spot news events have all helped build credibility.

Challenges To Coordination

While most broadcast licensees play by the Rules, employ competent engineers and operators, and respect fellow broadcasters rights, some do not. Such broadcasters are guilty of at least five deadly and not so deadly sins. Here are those sins and a catalog of syndromes documented by the Author over a 16 year period of involvement with the process:

<u>Arrogance</u>
"They'll Never Find Me"
"Who Cares About The Laws of Physics"
"We've Always Done It This Way"
"It's My Frequency"
"We're Bigger Than You Are"

Ignorance

"Sorry, I'm Just A TV Repairman"
"I Did Not Know That"
"We've Always Done It This Way"
"It's My Frequency" [12]
"Committee? What Committee?"
"I Checked the Database, So I'm OK"
"The Rental Company Said It Would Be OK"

Societal Breakdown

"The News Director Said 'Do It' "
"The FCC Doesn't Care" [13]
"The End Justifies The Means"

Miscommunications And Assumptions

"I Guess I Wrote It Down Wrong"
"Well, That's What I was Told To Do"
"I Did It That Way Last Year"
"A Told B and C Told Me"

Honest Breakdowns and Honest Apologies

"We're Sorry and Well Get It Fixed ASAP"
"The Manufacturer Didn't Tell Us"
"What Service Bulletin?"

Connect The Dots

Volunteer coordinators who are on the receiving end of such responses to complaints by other licensees should use the following procedure to help all the parties involved resolve conflicts:

1. Conduct a finding of fact.
2. Schedule an arbitration session with all parties.
3. Suggest possible solutions if necessary.
4. Test possible solution(s) if called for.
5. Allow a reasonable implementation period.
6. Request timely reports on implementation.
7. Has conflict been resolved?
8. Success!
9. Congratulations!
10. Write the final report.
11. Wait for the next one to come up.

THE CASE FOR THE CASE STUDY

Why Is Case Study Helpful to Coordination?

Local volunteers might learn something from law schools such as Harvard and Yale that use the case study method. This learning tool helps to firmly implant key issues, laws, rules, and principles in budding legal minds. Three cases are presented here in "sanitized" form to protect the innocent:

Case Study One

Background

Station "A" had been using a traffic service for many years. The traffic service held the Remote Pickup (RPU) license for airborne reports. Station "A" contracted with a new traffic service company to provide its reports. The new company obtained an RPU license under the same terms. [14] Station "B" had been licensed on the same channel under an old sharing agreement. Under that agreement, "B" used the channel infrequently for field coordination as did several other stations on this channel when the agreement was first struck. Station "B" began to use the channel for remote broadcasts on a regular basis. Station "A" then decided to add the channel used for airborne reports to its existing system license. "A" said they wanted to have more control over RPU equipment used for their traffic reports. Station "B" claimed that Station "A" was interfering with their remote operations. The coordination committee hosted an arbitration meeting. Among their claims, Station "B" said they were licensed on the channel first and their remote use had a higher priority under Part 74 rules than traffic reports. Station "A" claimed that they had a longer continued use of the channel and that the nature of their activity had equal priority. The local coordination group suggested several alternatives to Station "B" at the arbitration meeting held with all parties present. Station "B" felt, as the injured party, that Station "A" should purchase new equipment for them, or vacate the channel. Station "A" was unwilling to vacate the channel. Station "A" stated that it had worked out a sharing agreement so another client of the traffic service could use the channel for their reports when "A" was not doing airborne reports. Several weeks after the arbitration meeting, Station "B" filed a Petition To Deny the license application of Station "A" with the FCC.

FCC involvement

The FCC, after reviewing submissions from attorneys for "A" and "B", found that Station "A" had acted in good faith and that the facts that "A" and the coordinating entity submitted were substantially correct. The FCC asked the coordinating committee to hold another arbitration session and report back to the FCC on what happened.

Resolution

After the arbitration session, Station "B" agreed to apply for a modification to their RPU license after reaching agreement with another station (Station "C") occupying another channel where terrain and distance would protect both parties. After testing, Station "B" found that their remotes worked without interference. Station "C" received no interference.

Morals And Lessons

FCC intervention could have been avoided. The FCC supported the local committee and the local coordination process.

Legal involvement (and costs) could have been avoided. Committee suggestions for resolution were ultimately implemented.

Case Study Two

Background

Station "D" began to experience daily interference to its remote pickup operations. After utilizing direction finding techniques, the source was localized to a RPU repeater at another hilltop site that had been installed without a ferrite circulator, reject load, and properly tuned band pass/band reject filters. Station "D" contacted the engineering department of Station "E". After the problem was explained, Station "E" claimed that they purchased the repeater from a vendor who sold it to them without telling them that a circulator and reject load would prevent RF mixing in the output stage of the transmitter. Station "E" 's engineer was unaware of the high potential that spurious emissions would be produced without such devices.

FCC Involvement

None!

Local Committee Involvement

None!

Resolution

Station "E" purchased proper filtering accessories. Station "E" was pleasantly surprised that their system worked better. They were even able to cut back their RPU transmitter power. Station "D" reported that the interference had disappeared.

Morals And Lessons

Not all broadcast engineers are up to speed on two-way basics. When all parties approach problems with open minds and good will, problems can sometimes be solved. It is also rewarding to help educate others.

Case Study Three

Background

After years of status quo in the 2 GHz band of a Top 100 market, long standing ENG users were approached, through their local coordinating committee, to accommodate additional users who announced they were going to begin ENG operations. All but one of the existing stations agreed to a "Home Channel Plan", a method to bring about real time sharing that was pioneered in Washington, DC and successfully used under difficult and congested conditions during the 1984 Los Angeles Olympics. The hold out station refused to cooperate. Its engineering department stated that their news department controlled all ENG decision making. Further, the news director would not tolerate any circumstance where he would yield use of " their" ENG channel.

FCC Involvement

None

Local Committee Involvement

Behind the scenes!

Resolution

The coordinating committee worked behind the scenes with the local news directors (not the engineers) to reach an accommodation with all stations. The local committee obtained signatures on a local plan that acknowledged equal access for all users, protected the "rights" of all users, and laid out ground rules that were as fair as possible to all parties. After more than two years, the news director of the seventh station was the final signatory to the agreement.

CONCLUSIONS

The FCC leaves it up to Part 74 licensees to implement local spectrum use policies. Local voluntary coordination has is the engine that puts Part 74 policies for local coordination into practice. This effort embodies more than technical expertise. Political, social, and economic considerations come into play. While there is much support from the SBE for this vital local effort, more is needed from all licensees. Cooperation, proper operator training, and assistance with local coordination expenses are needed. Licensees owe a great deal to volunteer coordinators. They are our first and best line of defense against interference. It is to these individuals that this paper is dedicated.

FOOTNOTES

[1] The author has had many discussions on this topic with FCC Field Enforcement Bureau personnel in many parts of the country. Safety-related missions, ship board radio inspections, and searches for pirate transmitters in the AM and FM bands are some of the resource-hungry tasks FCC Field Office engineers must perform. Amateurs and broadcasters usually have to do their own detective work and careful documentation of evidence in order for the FCC to take action. The chances for rapid and effective FCC action increase in direct proportion to the amount and quality of such detective work.

[2] Part 74.402, the first Rules change to split the 450 band, became effective on November 22, 1976.

[3] Coordination is mentioned in two places in Part 74. The primary reference is under 74.24(g). It states that "The Part 74 licensee of this chapter, prior to operating pursuant to the provisions of this section, shall for the intended location or area of operations, notify the appropriate frequency coordinating committee or any licensee(s) assigned the use of the proposed

operating frequency....information on active frequency coordination committees may be obtained by contacting the Auxiliary Services Branch..." The Society of Broadcast Engineers (SBE) compiled the original list of local coordination groups for the FCC, and updates it quarterly. Section 74.403(a) indirectly refers to the local coordination process without mentioning it by name.

[4] Part 74.403 is entitled "Frequency Selection To Avoid Interference". It encourages licensees to "endeavor to select frequencies or schedule operation in such a manner as to avoid mutual interference." If a conflict is reported to the FCC, it further states, "...the Commission shall be notified and it will specify the frequency or frequencies on which each station is to be operated." The implication is clear. A local solution is to be preferred over an FCC solution from Washington.

[5] Licensees should pay close attention to this process since Form 313 asks if a local coordination entity has been contacted.

[6] Short term operation is governed by Part 74.24 of the Rules.

[7] The five priorities apply to all frequencies except certain channels and certain conditions specified under 74.402(a)(3); (7), and (8):

(1) Communications during an emergency or pending emergency directly related to safety of life or property.

(2) Program material to be broadcast.

(3) Cues, orders, and other related communications immediately necessary to the accomplishment of this broadcast.

(4) Operational communications.

(5) Tests or drills to check the performance of stand-by or emergency circuits.

[8] The Forward to the SBE Canons reads in part:
Honesty, justice, and courtesy form a moral philosophy when associated with mutual interest between human beings. This constitutes the foundation of ethics... Broadcast Engineers will strive to be fair, tolerant, and open minded.

[9] SBE works with the NAB on other matters that affect Part 74. SBE, NAB, the National Cable Television Association (NCTA), NBC, CBS, ABC, Turner and Group W formed an all-industry council to address issues that affect all authorized licensees.

[10] Section 74.461(b) specifies that airborne operations will normally be limited to a maximum transmitter output power (TPO) of 15 watts. Licenses for higher power will be issued only "upon an adequate engineering showing of need, and of the procedures taken to avoid harmful interference to other licensees." The power for most fixed and rotary wing traffic reporting can be far less than 5 watts TPO if good engineering practices are observed.

[11] Part 74.24(g), governing Short Term Operation, contains language that gives credence and a form of "blessing" to the local coordination process by stating that licensees operating under this section should "notify the appropriate frequency committee". The section goes on to state that information on active committees can be obtained directly from the FCC's Auxiliary Service Branch. This recognition is somewhat tempered by other language in this section.

[12] Also listed under <u>Ignorance</u> to be distinguished from this statement being made out of <u>Arrogance</u>. Part 74.402(e) outlines in detail that frequencies are assigned on a non-exclusive basis, and "the same frequency or frequencies may be assigned to other licensees in the same area."

[13] This statement has been heard in large and small markets.

[14] Many entities supplying airborne traffic reports have obtained Part 74 licenses under its network provisions. Part 74.2 defines a broadcast network-entity as "...an organization which produces programs available for simultaneous transmission by 10 or more affiliated broadcast stations and having distribution facilities or circuits available to such affiliated stations at least 12 hours each day."

Acknowledgments

Along with all the local volunteers who make Part 74 work, the author wishes to thank the following individuals for their time and help on this paper:

Mr. Dane Ericksen, P.E., of Hammett & Edison, Inc., Consulting Engineers, Chairman, SBE FCC Liaison Committee.

Attorney Chris Imlay; Booth Freret & Imlay, SBE Counsel

Mr. Paul Lentz who maintains the National Frequency Coordinator List for the SBE

Mr. John Russell, Chairman, Southern California Frequency Coordinating Committee

SBE DAY
DESIGNING A SERIAL DIGITAL PLANT

Tuesday, April 20, 1993

Moderator:
Bob Paulson, AVP Communication, Westborough, Massachusetts

SERIAL DIGITAL: THE ROAD TO OUR ALL-DIGITAL FUTURE
C. Robert Paulson
AVP Communication
Westborough, Massachusetts

***DIGITAL IN AN ANALOG ENVIRONMENT**
Robin Wilson
Grass Valley Group
Grass Valley, California

ERROR DETECTION IN SERIAL DIGITAL SYSTEMS
David K. Fibush
Tektronix, Inc.
Beaverton, Oregon

***BIT-BY-BIT: REAL WORLD EXPERIENCE IN DESIGNING SERIAL DIGITAL SYSTEMS**
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***SYSTEMS MAINTENANCE: A NEW ROLE FOR THE BROADCAST ENGINEER**
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SERIAL DIGITAL: THE ROAD TO OUR ALL-DIGITAL FUTURE

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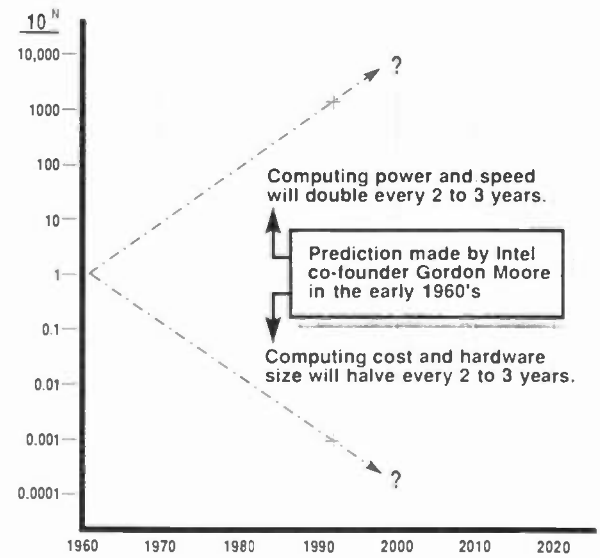
ABSTRACT

Computer pioneer Gordon Moore first articulated the economic and technical benefits of digital solid state technology in product design in the early 1960s, when he co-founded chip maker Intel Corporation. The computer and telecommunications industries have since then constantly fulfilled his "Moore's Law" prediction of doubled performance and halved costs at least every two to three years.

Meanwhile, the television industry has disconnectedly moved from its all-analog origins to "islands of digital in an analog sea of NTSC artifacts, connected by coax tunnels", inside the plant, and "aqueducts of analog in a 3-D global maze of digital satellites and fibers", outside the plant. Serial digital technology, exploding in performance while shrivelling in cost per Moore's Law, provides the broadcast industry a hope for survival in the all-digital environment in which the other two industries have immersed us.



Semi-log plot of "Moore's Law"
(What if the law had applied to Mercedes' cost/performance?)

THE SERIAL/PARALLEL DILEMMA

Analog is serial and parallel

Creation, transmission, processing and storage of analog video and audio were, are and ever will be serial processes. However, they are carried out in totally separate and different hardware systems, whose products must interconnected by totally separate, different, parallel transmission circuits. Wired and wireless transportation of video and audio signals are both made complicated and oft-times nearly impossible by legions of competing, and usually incompatible, continentally-originated and fought for standards and practices.

Inside the plant, video, audio, intercom and ancillary data signal distribution demands have spawned generations of ever-larger, ever more expensive multi-level routing switchers. A "baker's dozen" of analog television recorder formats continue to proliferate to provide a means of integrally and time-synchronously storing associated video, audio and data magnetically on one piece of recording medium. Hardware for *separately* storing video, audio and data boost the number of recording formats found in a television broadcast or production facility to well upwards of 50, not counting film or paper.

In today's television production or broadcasting plant, it's not uncommon to find operational needs for six or more levels of optional ganged and breakaway routing among or between some suites, systems, control consoles and storage systems. These always proliferating requirements can generally only be solved, expensively, by installing a hierarchy of routers with expandable minimum switching levels.

The equipment being interconnected by these routers still looks pretty much the same as it did twenty years ago, albeit it's smaller. Inside all that equipment, however, violent revolutions have been taking place during those twenty years. Few of them have any screwdriver slots for insertion of tweakers. That's because those traditional old analog cameras, recorders, transmitters, monitors, amplifiers, et al, are now first and foremost computer systems, most with a multiplicity of microprocessor chips. Operating and maintaining them requires computer literacy. Modifying their analog signal performance requires manual typing skill and computer logical thinking skills.

Digital is parallel and serial

Since the mid 1980s, new generations of these products have been designed to process video and audio input signals entirely in the digital domain. "DSFX" (Digital Special Effects) black boxes came first, to create, geometrically manipulate, combine and adjust still and motion video images, and store them on proprietary, non-interchangeable magnetic disk peripherals. They operate, as all computers do, in bit-parallel, word (Byte)-serial fashion. Word creation (sampling) rates are well above twice the highest signal frequency, the minimum specified by "the Nyquist theory." These "clock rates" range upward from 10 to 75 MHz.

Digital component and then digital composite (digitized NTSC) television tape recorders were the first product evolution. They accept bit-parallel/word serial outputs from the DSFX black boxes, with transportation accomplished over monstrous twisted pair cable terminated in monstrous 25-pin connectors. Television cameras are the latest product to join the parade to all-digital systems, in designs which digitize the R, G and B signals at the imaging chip preamp output.

Bit-parallel digital cable size and cost and inconvenience, and equipment separation limits of 50 meters of cable, are the practical drawback to the creation of all-digital television facilities. A second drawback, equally vexing and often unacceptable, is creative.

A "word" in computerese has always been eight bits long, defined as a "Byte." That word length is adequate to identify 128 ASCII (American Standard Code for Information Interchange) characters using seven bits. In video image creation and processing, however, eight bits is *not* enough to eliminate the "contouring" digital imaging artifact.

Eight bit resolution divides a video signal's peak to peak amplitude into 256 levels. Digital multiplication, rounding (the digital equivalent of analog signal resistive mixing), and re-recording, results in the creation of "rings" (like on a coffee table) demarking two levels of originally flat field colors. Ten bit resolution (1024 amplitude levels) pushes the detection of the contouring artifact down some more processing generations. Some DSFX boxes process signals internally at much higher bit resolutions to delay the onset of contouring.

And there, in a basket of rather large nutshells, is the substance of the serial/parallel dilemma that began to emerge in the late 1980s. Since then, signal resolution has moved from eight to ten bits and will continue upward. Sampling clock rates will continue to increase proportionally with demands for higher video resolutions. These variables work together to decrease the already onerous length limitation of massive bit parallel cables and connectors.

Up to now, no signal-transparent serial digital routing system has been available for signal transportation throughout a television plant. Thus, early 1990s television plants have become *"islands of digital in an analog sea of NTSC artifacts, connected by coax tunnels."* Existing analog routers and 75-ohm unbalanced coax cable networks require signal conversion from parallel digital components to serial analog composite before each transmission.

THE COPPER/GLASS ISSUE

Coaxial cable shortcomings

High quality coax cable is adequate for point-to-point transportation of a D-2/D-3 serial digital signal over about 1000 feet (300 meters).

However, a recent SMPTE paper by Leitch VP of Product Development Robert Proulx[1] cites the "cliff effect" of coax at transmission lengths approaching 400 meters. A test at a 144 Mbps bit rate showed zero bit errors at 370 meters, occasional errors at 380 meters, and system crash at 390 meters! The distance to the edge of the cliff is an inverse function of the the transmission bit rate.

It is obviously a "Given" that future progress toward the realization of all-digital television plants is tightly tied to the creation of new-concept routing and distribution systems. Hoping for lower-loss/wider bandwidth coax cable and wider bandwidth analog routing switcher developments is a head-in-the-sand attitude.

Fiber is the answer -- partly

Fiber-based transmission systems debuted in 1980 at the Lake Placid New York Winter Olympics venue installed by ABC Television. First-generation analog systems, carrying AM modulated NTSC and an FM subcarrier for audio, designed, installed and nursed by a GVG crew led by Birney Dayton, provided back hauls to the downtown operations center over many kilometers. (Triax cables with custom-designed repeaters designed by Philips also made their debut and swan song, permitting camera head/CCU separations of 20,000 feet).

Fiber-based analog systems for bi-directionally transmitting FM-modulated NTSC video, and ancillary multiplexed audio, intercom and data channels, are now available from several vendors. Digital-domain systems began to appear in the late 1980s. Analog systems designed to operate on Multi Mode fiber offer transmission ranges up to 20 to 25 kilometers at under $10,000 prices per link. Single Mode systems offer transmission ranges up to 40 kilometers and more at well upwards of $10,000.

High speed digital fiber-based systems expected to be debuted at NAB 1993 provide for transmission of many multiplexed "digital NTSC"/multiple program audio (D-2/D-3) channels over Single Mode fiber circuits. Transmission speeds of 1.2 and even 2.4 Gbps will be utilized. Transmission link costs are expected to be in the $6000 to perhaps $10,000 range, depending on the number of channels installed.

Proliferating serial digital interface needs

The uncompressed video/audio serial interface bit rates for the television tape recording formats in use now and looming in our future range from "sub T-1" (64 to 784 kbps), to 140 Mbps for "digitized NTSC" (SMPTE D-2/D-3), to approximately 1.2 Gbps for the forthcoming wide-screen/high-resolution broadcast standard. New requirements to serially transport signals in this nominal 20,000:1 speed range come from five sources. (1) Production and postproduction houses are committed to shooting and editing at "full Bandwidth" (1.2 Gbps). (2) However, the FCC has indicated that the eventually approved ATV signal will be digital, broadcast over existing 6 MHz UHF channels at a maximum of 20 Mbps.

(3) Broadcasters are now investigating "lossless" compression techniques (the original signal can be decompressed to its exact original form) that would allow transmission of "several" channels within the existing 6 MHz channels. (4) Cablecasters are moving pell mell toward systems for cable-transporting scores of compressed video/audio television programs over the same 6 MHz channel bandwidth. (5) Transmission speeds for severely compressed television signals deemed adequate for videoconferencing and distance education (and interactive computer games, etc. ??) range downward from 1.5 Mbps to 64 kbps.

Fiber link and switcher versatility needs

Fiber-based serial digital transmission links and serial digital crosspoint routers designed to operate at only one speed in this 20,000:1 transmission speed range are rather like a laser disk player that only reads one format. Who knows what new format s and transmission speeds in this in this spectrum will obsolete D-1, D-2, D-3, D-X, "sub T-1," DS-3 or any other existing digital recording or transmission format?

AVP Communication research has identified two criteria for "future-proofing" a television plant:

(1) - ***Fiber-based*** serial digital routing and distribution systems are mandatory; and

(2) - Transmission links ***must be*** recording format and transmission speed independent.

The NAB '93 "Missing Link" solution

In the course of a decade of research and writing to define future needs for format-independent serial digital fiber transmission links (Reference 2 et seq), AVP has both worked with and kept track of many fine companies attempting to find a profitable niche in the fiber transmission systems marketplace. At NAB '93, two such companies, MERET Optical Communications,Inc., Santa Monica California and PESA Switching Systems, Huntsville Alabama, will be debuting products meeting the "future-proof" criteria. Merger of the two products into an "OmniSpeed™ Integrated Routing Switcher and Fiber Distribution System" creates the "Missing Link" that enables system designers to move immediately to the layout of adaptable all-digital television plants.

MERET is debuting at NAB 1993 a serial digital fiber transmission link which operates at any throughput rate from ***5 through 400 Mbps.*** The Company's designers say that this is the first of a family of transmission links that will span the transmission speed range from 64 kbps to 2.4 Gbps. The transmitter and receiver packages are miniscule, 1 1/2" square by 3 1/2" long including fiber and BNC connectors. The link price is deemed to be "Cheap!", including tiny power supplies. MERET engineers are ready to talk about laying out the tiny boards for building in to anybody's equipment. The link automatically locks to any incoming signal in the specified speed range without needs for sync or clocking signals or tuning circuit tweaking.

PESA Switching Systems is debuting a serial digital routing switcher which accommodates serial digital signals at bit rates up to 600 Mbps. It is considering offering a system design option in which customer-specified numbers of MERET transmitters and receivers are built into its rack unit. Re-clocking is provided for signals operating at SMPTE standard bit rates.

Prices for these products will prove that Moore's Law is alive and well at NAB '93. There is every indication that performance versatility and speed will continue to double, and size and cost halve, in the years ahead.

The NAB '93 Missing Link Network

NAB attendees are well-known to be jaundiced in their suspicious viewing of "state of the art" new products. Accordingly, a "Missing Link" routing switcher and distribution network has been installed and is operating in several fiber-interconnected booths in the Las Vegas Convention Center South Hall. At deadline time for this paper, prospective Partners invited to operate on this switched serial digital network included PESA/Chyron Graphics, FOR-A, Ikegami, Magni Systems, NVision, Odetics Broadcast , Panasonic Broadcast and Vyvx National Video Network.

Each invited Partner brings a different kind of signal and transmission problem to the network. The transmission hardware and software solutions to the problems will involve the products of still other exhibitors. Some needs may not be satisfied before the 1993 NAB Exhibit opens. In those cases, the Missing Link network will highlight the additional solutions needed, before the all-digital serial television plant is as user-accommodating to plan and operate as early 1980s all-analog/all-NTSC plants.

INTERFACING THE DIGITAL ROAD OUTSIDE

Operation either in the way-back-when all-analog domain or the existing analog/digital islands and oceans environment has had two drawbacks. First, transporting signals to other locations in the production, postproduction and distribution infrastructure required the creation and use of *"aqueducts of analog in a 3-D global maze of digital satellites and fibers,"* Second, *NOISE BUILDUP* ("snow", streaks, decrease sharpness, color impurities) during signal re-recording and transmission forced the broadcast industry into two unattractive Modus Operandi (MOs).

One MO was expensive -- designing "full service" facilities in which every system in the broadcasting process was reachable via a routing switcher. That reduced noise buildup by lowering the number of re-recordings needed to reach the edited master goal. The other MO was expensive and time-consuming -- making recordings to provide physical transportation of signals to other locations in the outside global infrastructure.

There is no noise buildup from re-recording in digital (although compositing and re-recording in digital composite NTSC does inevitably produce "NTSC footprints" -- "creepie crawlies"). There is also no noise buildup from transporting signals around an electrically noisy plant, until the cliff effect takes over. Digital repeaters installed in the transmission line accommodate extended transmission distance needs.

These benefits of all-digital operation in plant suggest that there are substantial economic benefits awaiting by digitally interfacing the all-digital plant to the long-time all-digital global telecommunications industry infrastructure. One would evolve from recognition of the implications of the "Seven/24 syndrome" -- grouping expensive DSP equipment resources at locations where it could be kept in use seven days a week, 24 hours a day. The other would be the ability to work on various tasks of production and postproduction at different specialized facilities at the same time, and passing Work in Process (WIP) to other facilities instantly as completed.

Worthy objectives, these! But not realizable simply by poking digital fiber cables through the telephone plant walls. There are two obstacles to creation of the all-digital television world, one technical, the other regulatory. These will be elaborated on during the paper's presentation. They are also being proposed as topics in a "Designing the All-Digital Global Television World" engineering session at the September 1993 SBE Conference in Miami.

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ERROR DETECTION IN SERIAL DIGITAL SYSTEMS

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Abstract

Use of error measurement techniques provides a valuable extension to television system diagnostics. When errors occur it means there is a problem because there is no measurable rate of errors that is acceptable in a studio digital television system due to the nature of the transmission method and the visibility of the errors. The definition and measurement of errors in serial digital television signals must take into consideration the specific attributes of television equipment and the digital interconnection system. This paper compares the use of traditional bit error rate measurement techniques with a method designed to be part of the digital television signal.

Introduction

In our application of computers and digital techniques to automate many tasks we have grown to expect the hardware and software to report problems to an operator. This is particularly important as systems become larger, more complex and more sophisticated. It is reasonable to expect digital television systems to provide similar fault reporting capabilities. Television equipment has traditionally had varying levels of diagnostic capabilities. With the development of digital audio and video signals it is now possible to add a vital tool to system fault reporting, confirmation of signal integrity in studio interconnection systems. To understand and use this tool it is necessary to delve into the technology of digital error detection. The combination of the serial digital video interconnect method, the nature of television systems, and typical digital equipment design lead to the use of error measurement methods that are economical yet specialized for this application. Serial digital video operates in a basically noise free environment so traditional methods of measuring random error rates are not particularly useful. Examination of the overall system characteristics leads to the conclusion that a specialized burst error measurement system will provide the television studio engineer with the tool to monitor and evaluate error performance of serial digital video systems.

Definition of Errors

In the testing of in-studio transmission links and operational equipment a single error is defined as one data word whose digital value changes between the signal source and the measuring receiver. This is significantly different than the analog case where a range of received values could be considered to be correct. Application of this definition to transmission links is straight forward. If any digital data word values change between transmitter and receiver there is an error. For operational studio equipment the situation is a little more complicated and needs further definition.

Equipment such as routing switchers, distribution amplifiers and patch panels should, in general, not change the data carried by the signal hence the basic definition applies. However, when a routing switcher changes the source selection there will be a short disturbance. This should not be considered an error although the length of the disturbance is subject to evaluation and measurement. SMPTE Recommended Practice RP 168[1] specifies the line in the vertical interval where switching is to take place which allows error measurement equipment to ignore this acceptable error.

There are various studio equipments that would normally be expected to not change the signal. These would be such things as frame synchronizers with no proc amp controls, production switchers in a straight-through mode and digital VTRs in the E-to-E mode. All have the capability (and are likely to) replace all of the horizontal and part of the vertical blanking intervals. In general replacing part of the television signal will destroy the error-free integrity of the signal. This is true for both component and composite signals as there could be ancillary data in the replaced sections of the signal. There is an even stronger case for composite signals as the sync areas are not a strictly defined set of digits, hence may vary within the limits of the analog signal standard.

Considering the possible replacement of blanking areas by some equipment and knowing that the active picture sample locations are well defined by the various digital standards, it is possible to measure Active Picture Errors as distinct from Full Field Errors. The Active Picture Error concept takes care of blanking area replacements, however, there is a more insidious possibility based on the television design engineer's historical *right* to modify the blanking edges. Most digital standards were designed with digital blanking more narrow than analog blanking to ensure appropriate edge transitions in the analog domain. (Large amplitude edge transitions within one sample period caused by time truncating the blanking waveform can create out-of-band spectral components that can show as excessive ringing upon digital to analog conversion and filtering.) What happens in some equipment is that engineers have modified the samples representing the analog blanking edge in order to provide a desired transition, therefore, error measurements through such equipment even using the Active Picture concept will not work.

With VTRs there is a further limit to measurement of errors. The original full bit rate VTR formats (D-1, D-2 and D-3) are designed with significant amounts of forward error correction and work very well. However it is the nature of the tape recording and reproduction process that some errors will not be corrected. Generally the error correction system identifies the uncorrected data and sophisticated error concealment systems can be applied to make a virtually perfect picture. The systems are so good that tens of generations are possible with no human-noticeable defect. Although the error concealment is excellent it is almost certain that error measurement systems (that by definition require perfect data reproduction) would find errors in many completely acceptable fields. New VTR formats that use bit rate reduction will have an additional potential source of small but acceptable data value changes. The data compression and decompressions schemes are not absolutely lossless resulting in some minor modification of the data. Therefore, all error measurement methods external to a digital VTR are not meaningful because of potential blanking edge adjustment, error concealment methods and, in some equipment, the use of data rate reduction. The good news is that various types of error rate and concealment rate data are available inside the VTR and standards for reporting the information are being developed.

Quantifying Errors

Most engineers are familiar with the concept of Bit Error Rate (BER) which is the ratio of bits in error to total bits. As an example, the 10-bit digital component data rate is 270 Mb/s. If there were one error per frame the BER would be $30/(270 \times 10^6) = 1.11 \times 10^{-7}$ for 525 line systems or $25/(270 \times 10^6) = 0.93 \times 10^{-7}$ for 625 line systems. Table 1 shows BER for one error over different lengths of time for various television systems. BER is a useful measure of system performance where the signal-to-noise ratio at the receiver is such that noise produced random errors occur.

As part of the serial digital interconnect system scrambling is used to lower dc content of the signal and provide sufficient zero crossings for reliable clock recovery. It is the nature of the descrambler that a single bit error will absolutely cause an error in two words (samples) and will have a 50% probability of the error in one of the words being in the most, or next to the most, significant bit. Therefore an error rate of 1 error/frame will be noticeable by a reasonably patient observer. If it is noticeable it is unacceptable but it is even more unacceptable because of what it tells us about the operation of the serial transmission system. Nature of the Studio Digital Video Transmission Systems

Table 1. Error Frequency and Bit Error Rates			
Time Between	NTSC 143 Mb/s	PAL 177 Mb/s	Component Errors 270 Mb/s
1 television frame	2×10^{-7}	2×10^{-7}	1×10^{-7}
1 second	7×10^{-9}	6×10^{-9}	4×10^{-9}
1 minute	1×10^{-10}	9×10^{-11}	6×10^{-11}
1 hour	2×10^{-12}	2×10^{-12}	1×10^{-12}
1 day	8×10^{-14}	7×10^{-14}	4×10^{-14}
1 week	1×10^{-14}	9×10^{-15}	6×10^{-15}
1 month	3×10^{-15}	2×10^{-15}	1×10^{-15}
1 year	2×10^{-16}	2×10^{-16}	1×10^{-16}
1 decade	2×10^{-17}	2×10^{-17}	1×10^{-17}
1 century	2×10^{-18}	2×10^{-18}	1×10^{-18}

Specifications for sources of digital video signals are defined by SMPTE 259M[2]. Although the specifications do not include a signal-to-noise ratio (SNR) typical values would be 40 dB or greater at the transmitter. Errors will occur if the SNR at some location in the system reaches a low enough value, generally in the vicinity of 20 dB. Figure 1 is a block diagram of the basic serial transmitter and receiver system. An intuitive method of testing the serial system is to add cable and that is a straight forward method of lowering the SNR. Since coax itself is not a significant noise source it is the noise figure of the receiver that will determine the operating SNR. Assuming an automatic equalizer in the receiver, as more cable is added, eventually the signal level due to coax attenuation will cause the SNR in the receiver to be such that errors occur.

Based on the scrambled NRZI channel code used and assuming gaussian distributed noise a calculation using the error-function gives the theoretical values[3] shown in Table 2. The calibration point for this calculation is based on the capabilities of the serial digital interface. In the proposed serial digital standard it is stated that the expected operational distance is through a length of coax which attenuates a frequency of 1/2 the clock rate by *up to* 30 dB. That is, receivers may be designed with less or more capability but the 30 dB value is considered to be realizable. The data in Table 2 for NTSC serial digital transmission shows that a 4.7 dB increase in SNR changes the result from 1 error per frame to 1 error per century. For NTSC the calibration point for the calculation is 400 meters of Belden 8281 coax. (Various types of coax can be used provided they have a frequency response reasonably meeting the $1/\sqrt{f}$ characteristic described in the proposed standard. It is the 30 dB of loss point that is important.)

This same theoretical data can be expressed in a different manner to show error rates as a function of cable length seen in Table 3 and shown graphically in Figure 2. The graph makes it very apparent that there is a sharp knee in the cable length verses error rate curve. 18 additional meters of cable (5% of the total) moves operation from the knee to completely unacceptable while 50 less meters of cable (12% of the total length) moves operation to a reasonably safe, 1 error/month. Similar results will be obtained for other standards where the calibration point for the calculation is 360 meters for PAL and 290 meters for component. Cable lengths for headroom will scale proportionally.

Good engineering practice would suggest a 6 dB margin or 80 meters of cable, hence an maximum operating length of about 320 meters in an NTSC system where the knee of the curve is at 400 meters. At that operating level there should never be any errors (at least not in our life time).

Table 2. Error Rate as a Function of SNR for Digital NTSC			
Time Between Errors	BER	SNR (dB)	SNR (volts ratio)
1 microsecond	7×10^{-3}	10.8	12
1 millisecond	7×10^{-6}	15.8	38
1 television frame	2×10^{-7}	17.1	51
1 second	7×10^{-9}	18.1	64
1 minute	1×10^{-10}	19.0	80
1 day	8×10^{-14}	20.4	109
1 month	3×10^{-15}	20.9	122
1 century	2×10^{-18}	21.8	150

Table 3. Error Rate as a Function of Cable Length Using 8281 Coax in for Serial Digital NTSC			
Time Between Errors	BER	Cable Length (meters)	Attenuation (dB) at 1/2 Clock Freq
1 microsecond	7×10^{-3}	484	36.3
1 millisecond	7×10^{-6}	418	31.3
1 television frame	2×10^{-7}	400	30.0
1 second	7×10^{-9}	387	29.0
1 minute	1×10^{-10}	374	28.1
1 day	8×10^{-14}	356	26.7
1 month	3×10^{-15}	350	26.2
1 century	2×10^{-18}	338	25.3

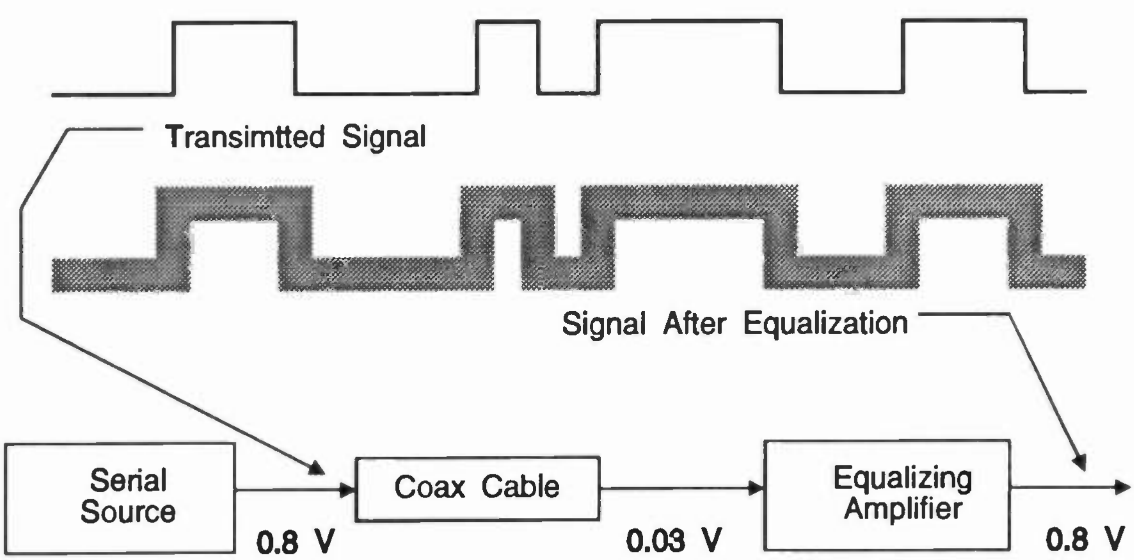


Figure 1. Basic serial transmitter/receiver system

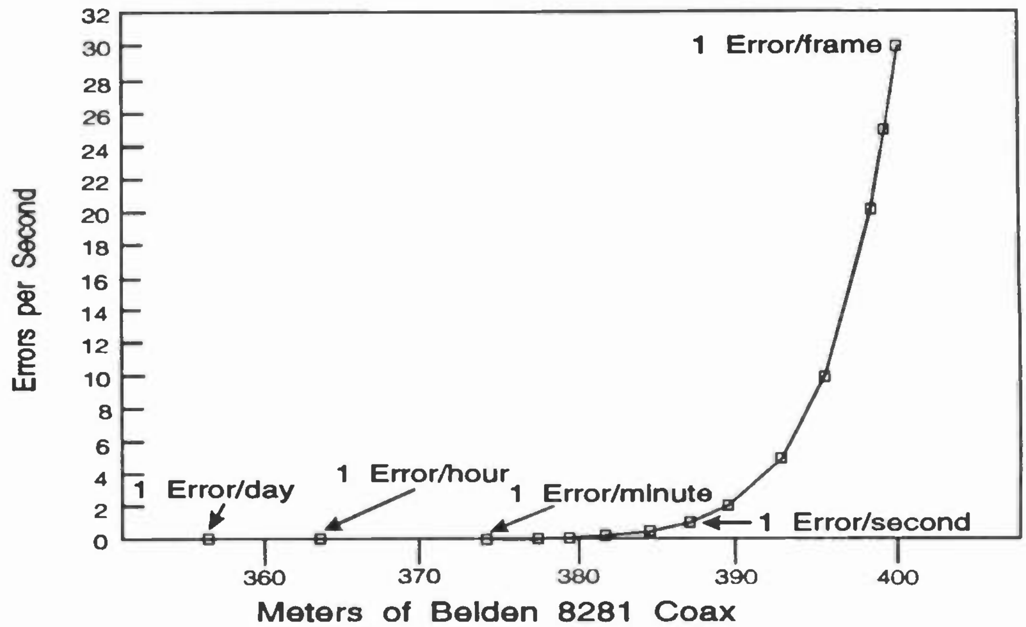


Figure 2. Error rate as a function of cable length for NTSC

Practical systems will include equipment that does not necessarily completely reconstitute the signal in terms of SNR. That is, sending the signal through a distribution amplifier or routing switcher may result in a completely useful but not completely standard signal to be sent to a receiving device. The *non-standardness* could be both jitter and noise but the sharp knee characteristic of the system would remain, occurring at a different amount of signal attenuation. Use of properly equalized and re-clocked distribution and routing equipment at intervals with adequate headroom will provide virtually unlimited total transmission distances.

Measuring Bit Error Rates

BER measurements may be made directly using equipment specifically designed for that purpose. Unfortunately, in a properly operating system, say with 6 dB of headroom, there will be no BER to measure. This is because the serial digital television system generally operates in an environment that is free from random errors. If errors are never going to occur as implied by Table 2 with 6 dB of headroom, what is there to measure? A more common problem will be burst errors due to some sort of interfering signal such as a noise spike that occurs at intermittent intervals spaced far apart in time. Another source could be crosstalk that might come and go depending on what other signals are being used at a particular time. Also there is the poor electrical connection at an interface that would cause noise only when it is mechanically disturbed.

Because of the intermittent nature of burst errors the data recording and communications engineers have defined another error measurement, the Errored Second. As an example, suppose a burst error causes 10,000 errors in two frames of video. A BER measurement made for 1 minute would indicate a 1×10^{-6} BER and a measurement made for 1 day would indicate a 8×10^{-9} BER. Whereas an errored second measurement could indicate that there was one second in error at a time 3 hours, 10 minutes and 5 seconds ago. The errored second method is clearly a more useful measurement in this case.

A significant advantage of errored seconds verses a straight BER measurement is that it is a better measure of *fitness for service* of links that are subject to burst errors. The serial digital video system fits this category because television images are greatly disturbed my momentary loss in synchronization. A BER measurement could give the same value for a single, large burst as it does for several shorter scattered bursts. But if several of the shorter bursts each result in momentary sync failure, the subjective effect will be more damage to the viewed picture than caused by the single burst. Errored seconds, and its inverse, error free seconds, do a good job of quantifying this.

For use in serial digital television systems there are several disadvantages to direct measurement of BER.

1. It must be an out-of-service test because traditional BER measurements use one of several defined pseudo random sequences at various bit rates.

2. None of the sequences are particularly similar to the serial digital video bit stream, hence, some television equipment will not process the test set bit patterns.

3. Consider a system operating with a reasonable 6 dB of SNR margin with respect to 1 error/frame. Errors due to random noise will occur once every hundred years or so. A BER measurement will not be possible.

4. BER measurements do not provide meaningful data when the system under test is basically noise free but potentially subject to burst errors.

5. Test sets for BER measurement are expensive considering their limited application in a television system.

An Error Measurement Method for Television

Tektronix has developed an error detection system for digital television signals(4) and has placed the technical details in the public domain to encourage other manufacturers to use the method. This has proven to be a sensitive and accurate way to determine if the system is operating correctly and it has been approved for standardization by SMPTE as Recommended Practice RP 165. Briefly, the EDH (Error Detection and Handling) concept is based on making CRC (Cyclic Redundancy Code) calculations for each field of video at the serializer as shown in Figure 3. Separate CRCs for the full field and active picture, along with status flags, are then sent with the other serial data through the transmission system. The CRCs are recalculated at the deserializer and, if not identical to the transmitted values, an error is indicated. Typical error detection data will be presented as errored seconds over a period of time, and time since the last errored second.

In normal operation of the serial digital interface there will be no errors to measure. What is of interest to the television engineer is the amount of headroom that is available in the system. That is, how much stressing could be added to the system before the knee of the error rate curve or *crash point* is reached. As an out-of-service test this can be determined by adding cable or other stressing method until the onset of errors. Since it is an out-of-service test either a BER test set or RP 165 could be used. There are, however, many advantages to using the RP 165 system.

1. The CRC data is part of the serial digital television signal thereby providing a meaningful measure of system performance.

2. RP 165 can be used as an in-service test to automatically and electronically pinpoint any system failures.

3. For out-of-service testing, RP 165 is sufficiently sensitive to accurately define the knee of the error rate curve during stressing tests.

4. Where there are errors present, RP 165 provides the information necessary to determine *errored seconds* which is more useful than bit error rate.

5. Facility is provided for measuring both full field and active picture errors. Optional status flags are also available to facilitate error reporting.

6. CRC calculation can be built into all serial transmitters and receivers for a very small incremental cost. With error information available from a variety of television equipment the results can then be routed to a central collection point for overall system diagnostics.

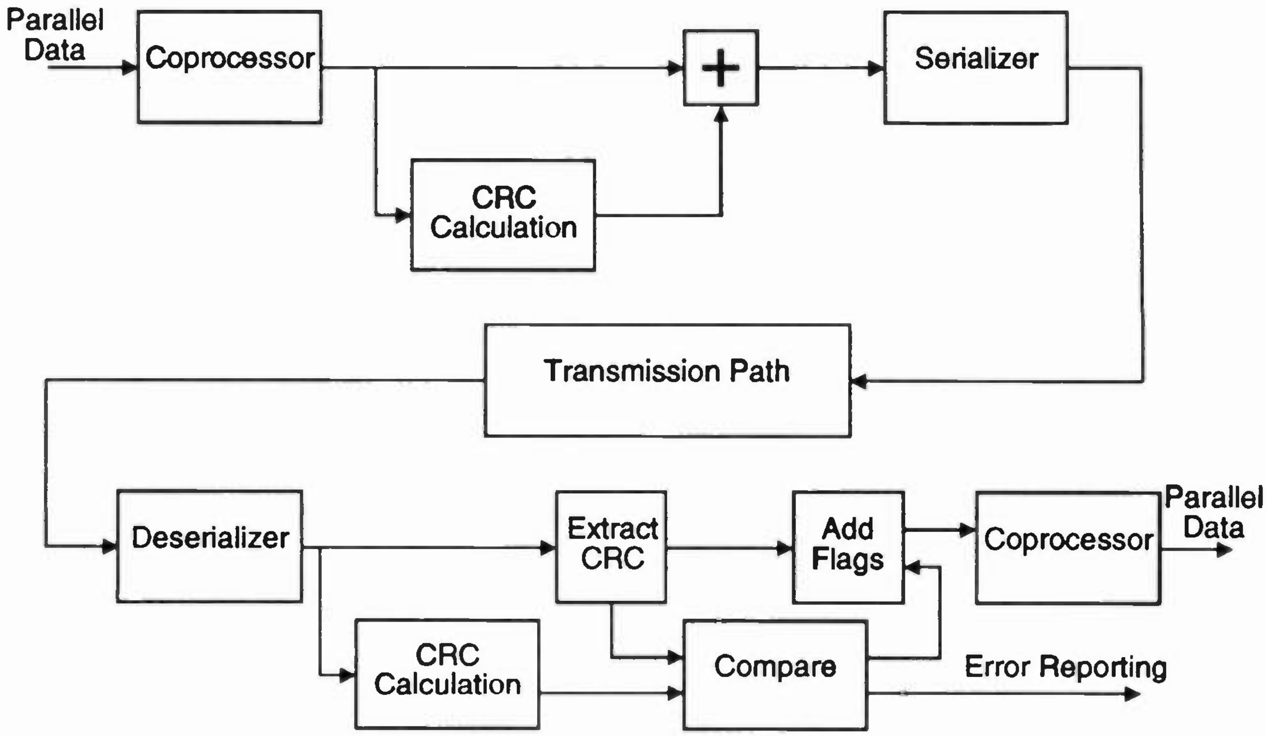


Figure 3. Error Detection and Handling - RP 165

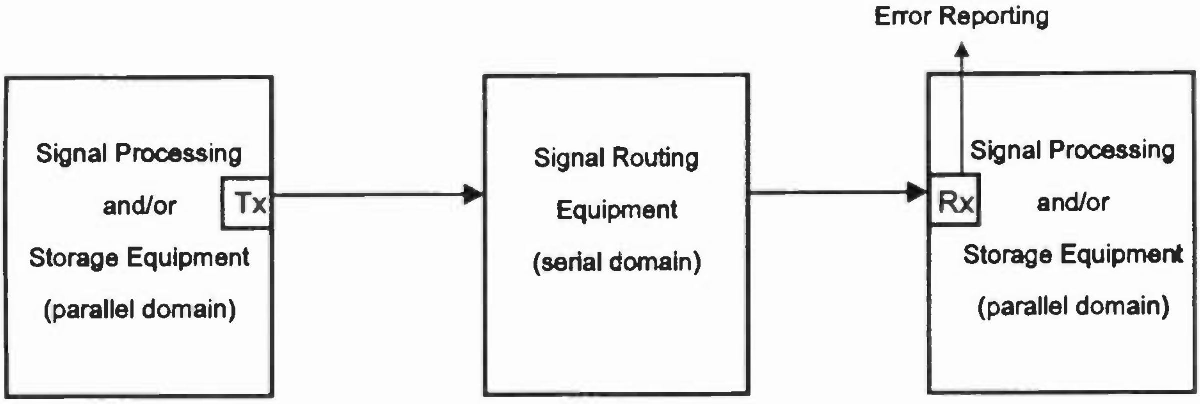
Composite digital systems already must have a coprocessor to handle special timing signals required in the serial data and not allowed in the parallel data(5). Adding basic CRC capabilities to the coprocessor costs close to nothing. For component systems which do not have the differences between serial and parallel, there is a small incremental cost which should be a fraction of a percent of the total cost in equipment where it is appropriate to use this method.

System Considerations

In a large serial digital installation the need for traditional waveform monitoring can be reduced by the systematic application of error detection methods. At signal source locations, such as, cameras, DVEs and VTRs, where operational controls can affect the program signal it will continue to be important to verify the key program signal parameters using waveform monitors with serial digital input capabilities. The results of certain technical operations, such as, embedded audio mux/demux, serial link transmission, and routing switcher I/O can be monitored with less sophisticated digital-only equipment provided the signal data integrity is verified. It is the function of the RP 165 system to provide that verification.

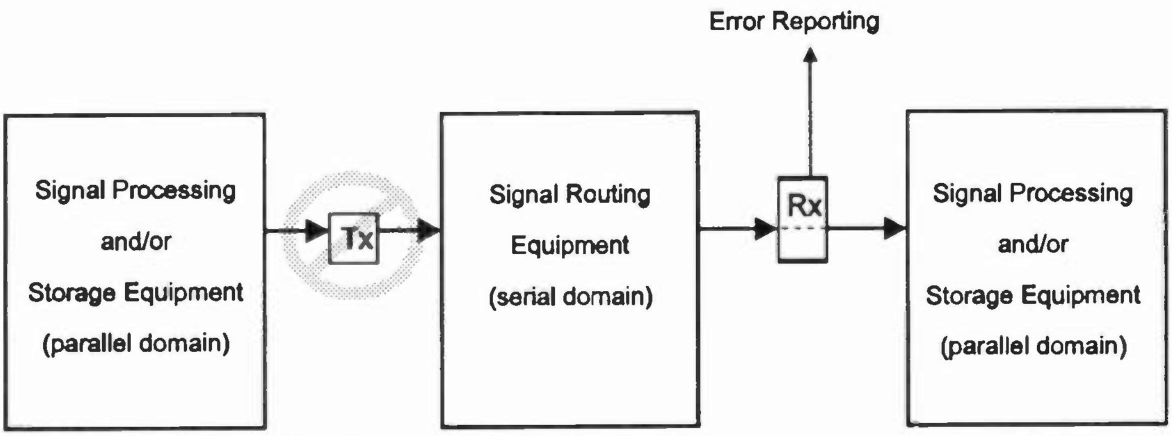
For equipment where the digital signal remains in the serial domain, such as routing switchers or distribution amplifiers, it is not economical to provide the CRC calculation function. Therefore, a basic error detection system will be implemented as shown in Figure 4. Equipment that processes the signal in the parallel domain, such as VTRs, DVEs or production switchers, should provide the transmit and receive CRC calculation necessary for error detection. That equipment can then report errors locally and/or to a central diagnostics computer. Routing switchers and other equipment operating in the serial domain will have their signal path integrity verified by the error detection system.

To emphasize the importance of including the CRC calculation in parallel domain processing equipment, consider the block diagram in Figure 5. Since the signal source does not have an internal CRC data generator one might consider a black box to provide that function. Unfortunately the cost of deserializing and reserializing makes the cost prohibitively high so



Tx = CRC calculation
Rx = CRC calculation and compare

Figure 4. Basic error detection system



Tx = Deserialize, CRC calculation, Reserialize
Rx = Monitor with active or passive loop-thru, CRC calculation and compare

Figure 5. Alternate error detection system (?)

such a system is not recommended. At a receiver which does not calculate the CRC it is possible, and not unreasonable, to provide a monitor with either passive or active loop-through which will determine the signal integrity. The drawback of this system is that it is the monitor receiver that is being used to process the signal not the actual destination equipment. It is possible that different receiver characteristics of the two pieces of equipment would provide misleading results.

To extend the idea of system diagnostics beyond simple digital data error detection it is necessary to provide an equipment fault reporting system. The SMPTE is presently working on a draft document for a single contact closure fault reporting system and discussions have begun on computer data communications protocols for more sophisticated fault reporting. Error detection is an important part of system diagnostics as shown in Figure 6. Information relating to signal transmission errors are combined with other equipment internal diagnostics to be sent to a central computer. A large routing switcher is large enough to economically support internal diagnostics. Using two serial receivers a polling of serial data errors could be implemented. The normal internal bus structure could be used to send serial data to a receiver to detect input errors and a special output bus could be used to ensure that no errors were created within the large serial domain routing switcher.

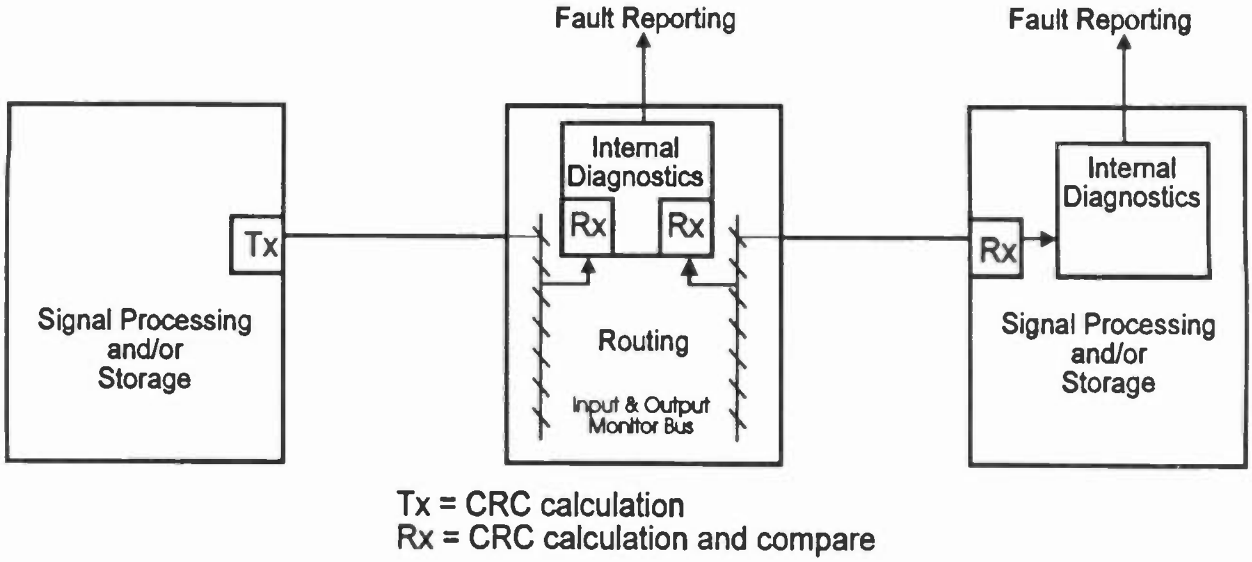


Figure 6. System Fault Reporting

Conclusion

No measurable error rate is acceptable due to the steepness of the error rate *knee.* Any errors (except for burst errors) indicate a dangerously low amount of system margin. Error measurements in digital systems are only effective where the transmission path or operational equipment does not alter the data of, at least, the active picture. Digital VTRs and signal processing equipment generally do alter the active picture, hence, their error performance can only be determined by internal means. Due to the crash point nature of digital systems and the likelihood of transient burst errors, typical bit error rate measurement methods are not particularly useful.

Just as we expect in a computer system, the digital television system should be able to tell us when it is broken and the location of the fault. A simple method of measuring errored seconds can be included in studio digital television systems to provide automatic in-service measurement of system failures as well as out-of-service headroom measurements.

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ROUTING SERIAL DIGITAL TELEVISION SIGNALS

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Televison production, post-production and broadcasting facilities of the 1980's were earlier described as consisting of "islands of digital in a sea of NTSC Artifacts." continued operation of "Moore's Law" (doubling digital capabilities and halving digital costs) is now providing digital hardware tools for transforming that NTSC sea into channels of serial digital flow. Digitized video and audio signals can readily be transported as a single high bit rate serial "data" stream. Fiber Optic serial digital transport systems provide economical, format-transparent, distance-independent transmission capabilities within a production, post-production or broadcast complex, and facilitate interfacing to outside plant.

INTRODUCTION

A substantial amount of information has been written about the relative merits of digital techniques and analog techniques used in the broadcast industry. Certainly the scene being viewed by a television camera is "analog" and the human eye that ultimately views the television picture is an "analog device." Prior to the early 1970's, virtually every piece of equipment involved in the production, processing, storage, transmission, and display of a television picture operated in the analog domain.

Digital circuitry and devices have characteristics that are very different from analog circuitry and devices. The characteristics that differentiate these two classes of devices and equipment are numerous and have been described in published literature. The first mass market for digital devices and circuitry was in the field of high speed computational equipment. The dramatic increase in the computational capability versus cost of digital devices that has occurred over time is well known to almost everyone. It was not surprising that digital equipment was first introduced into the broadcast industry for signal processing applications. In 1973, digital time-based correctors were introduced. This digital equipment rapidly evolved into field and frame synchronizers, digital special effects equipment, and character/graphic generators. These "digital signal processor" based devices represented the only digital equipment that was widely used in the broadcast industry until digital video recorders emerged in the late 1970's.

Throughout most of the 1980's the digital video recorder and the digital signal processing equipment remained as the only "digital islands in an otherwise all analog world of television production and post production." In recent years we have seen the introduction of additional digital equipment starting from television cameras with digital output all the way through to television monitors with digital inputs. The world of television production and post-production is rapidly changing to that of "an all digital world with an occasional island of analog equipment." As more and more digital equipment has been introduced into the broadcast industry the use of digital transmission techniques to interconnect this equipment has grown in popularity.

DIGITAL TRANSMISSION

The advantages of digital transmission are numerous with the primary advantages relating to transporting signals with no degradation of the performance parameters of signal to noise, differential phase, differential gain, etc. As always there is "no such thing as a free lunch" -- digital transmission systems can introduce bit errors in the transported signal and in multi-segment transmissions systems phase jitters must be considered in the design to assure proper transportation of the digitized video signals. Digital transmission has existed long enough for a reasonable set of digital video standards to be established. Table 1 delineates the standards that are the most relevant to the broadcast industry for both component and composite digital signals. Both

CCIR 601	Standard for 4:2:2 sampling
CCIR 656	Interfaces for digital component video signal in 525- and 625-line television systems
SMPTE T14.224	Serial digital interface for 10-bit 4:2:2 component and NTSC composite digital signals
SMPTE RP 125	Bit -parallel digital interface for component video signals
SMPTE 240M	Standard for 1125/60 high-definition production system
SMPTE 227M	Standard for recording digital television signals on magnetic tape in cassettes (D-1 for 526/60 video)
EBU Doc. Tech. 3252 E	Standard for recording digital television signals on magnetic tape in cassettes (D-1 for 626/50 video)
AMPEX	Digital format for video and audio tape recording of composite video signal using 19 mm type D-2 cassette

Table 1: Digital Video Standards for Broadcasting Production Applications

SYSTEM	SAMPLE RATE	WORD RATE	8-BIT DATA RATE	10-BIT DATA RATE
NTSC	14.32MHz	14.32MHz	111.5Mb/s	143.2Mb/s
PAL	17.7MHz	17.7MHz	141.6Mb/s	177Mb/s
CCIR 656	13.5MHz	27MHz	216Mb/s	270Mb/s
EDTV	18Mz	36MHz	288Mb/s	360Mb/s
EU95	72MHz	114MHz	1152Mb/s	1440Mb/s
SMPTE 240M	74.25MHz	148.5MHz	1188Mb/s	1485Mb/s

Table 2: Data Rate for Several Digital Video Interfaces[1]

Return loss	15 dB
Frequency range	5 MHz - 270 MHz
Peak-to-peak amplitude	800 mV ± 10%
D.C. offset	± 0.5 V
Rise and fall times (20% - 80%)	0.5 ns
Jitter	± 0.25 ns
Connector	BNC
Channel coding	Scrambled NRZI
Data word length	10 bits
Bit rate for 4:2:2 component	270 Mb/s
Bit rate for 4fsc composite NTSC	143 Mb/s
Bit rate for 625 line composite PAL (non standard)	177 Mb/s

Table 3: Basic Digital Serial Specifications (SMPTE T14.224)

parallel and serial digital formats are defined within the standards. In most applications today, serial digital video interfaces are used due to the savings in the cost of cabling the systems, the ease of routing serial digital signals, and the elimination of the need for delay equalization.The bit rate associated with the serial digital video signals for various standards is shown in Table 2 . Most of the standards define both 8 bit and 10 bit encoding of the luminance and the color signals. The serial bit data rates defined in these standards vary all the way from 111.5 Mb/s for 8 bit encoded NTSC signals all the way up to 1.485 Gb/s for the SMPTE standard for High Definition Television. A more complete summary of the key specifications defined in SMPTE standard T14.224 are shown in Table 3.

As mentioned above the two primary impairments that occur in digital transmission systems are the introduction of random errors and the phase jitter build up that occurs in multi-segment transmission systems. Of these two, the bit error rate is of primary concern in designing a digital production, post production or broadcast system. A bit error alters either the color or luminance signal for the particular pixel defined by the word containing the bit error. Bit errors once introduced, can not be corrected without the use of elaborate and expensive bit error correcting techniques. The preferred method for managing the bit error rate is to design each segment of the transmission system to have a bit error rate in the order of $1x10^{-13}$ (or one error per 10^{+13} second) bit errors per second or better. Table 4 defines the time between errors versus the bit error rates for various serial digital standards. Table 5 presents the theoretical error rate as a function of cable length for serial digital NTSC signals being transported over 8281 coaxial cable. This table shows one of the interesting characteristics of digital transmission specifically, that for cable lengths of less the 350 meters the bit error rate is literally unmeasurable, at 370 meters the bit error rate is just in the acceptable range, and by 390 meters the bit error rate is completely unacceptable. The implication of this table is that signal regeneration and repeatering is necessary every 350 meters along a coaxial/cable transmission path in order to achieve an acceptable bit error rate. However, recently introduced serial digital fiber optics links eliminate the need to regenerate the signals on longer transmission paths. Fiber optic systems can easily go 10 km or farther with acceptable bit error rates. Fiber optic systems have other advantages in terms of the physical size and cost of fiber versus that of coaxial cable. In the past, the optical transmitters and receivers necessary to use fiber optic cables for serial transmission of video systems have been quite expensive. As typically happens with solid state digital devices, the price of the photonics components which have the performance characteristics necessary for serial digital transmission links have been decreasing rapidly. Today a 10 km fiber optic link capable of carrying serial digital video signals can be built for a few thousand dollars.

Time between errors	NTSC 143 Mb/s	PAL 177 Mb/s	4:2:2 270Mb/s
1 TV frame	$2x10^{-7}$	$2x10^{-7}$	$1x10^{-7}$
1 second	$7x10^{-9}$	$6x10^{-9}$	$4x10^{-9}$
1 minute	$1x10^{-10}$	$9x10^{-11}$	$6x10^{-11}$
1 hour	$2x10^{-12}$	$2x10^{-12}$	$1x10^{-12}$
1 day	$8x10^{-14}$	$7x10^{-14}$	$4x10^{-14}$
1 week	$1x10^{-14}$	$9x10^{-15}$	$6x10^{-15}$
1 month	$3x10^{-15}$	$2x10^{-15}$	$1x10^{-15}$
1 year	$2x10^{-16}$	$2x10^{-16}$	$1x10^{-16}$
1 decade	$2x10^{-17}$	$2x10^{-17}$	$1x10^{-17}$
1 century	$2x10^{-18}$	$2x10^{-18}$	$1x10^{-18}$

Table 4: Error Frequency and Bit Error Rates[2]

Time between errors	BER	Cable Length (meters)	Attenution (dB) at Ω Clock Freq
1 microsecond	$7x10^{-3}$	483	36.3
1 millisecond	$7x10^{-6}$	417	31.3
1 TV frame	$2x10^{-7}$	400	30.0
1 second	$7x10^{-9}$	387	29.0
1 minute	$1x10^{-10}$	375	28.1
1 day	$8x10^{-14}$	367	27.5
1 month	$3x10^{-15}$	360	27.0
1 century	$2x10^{-18}$	348	26.1

Table 5: Theoretical Error Rate as a Function of Cable Length for Serial Digital NTSC Using 8281 Coax[2]

MIGRATION FROM ANALOG TO DIGITAL

The transition of video production, post production and broadcast from an all analog world to an all digital world seems to be a fait accompli. All-digital post production companies already exist and a substantial amount of digital equipment is currently in use in television production. Based on the testing that is currently being done on the broadcast of high definition television, it would appear that in the near future we will be broadcasting digital television signals. Transitioning from an analog production facility to a digital production facility does not necessarily mean ripping out all the old analog equipment and replacing it with all new digital equipment. Analog to digital and digital to analog conversion equipment exist that permit a "graceful" evolution path from an all analog world to eventually an all digital world.

The broadcast industry is not the first to be faced with the conversion from an analog to a digital world and by no means will be the last. The telephone system in the United States has been going through a similar transition for about the last twenty years. Prior to the 1970's the transmission and switching systems deployed throughout the United States were almost all analog. In the early 70's the telephone systems started deploying digital switches both in the expansion segment of their market and to replace fully depreciated and obsolete analog switching equipment. The digital switching equipment had numerous advantages over the older analog equipment, but it still had to interface into analog subscriber equipment and analog transmission equipment. Analog to digital and digital to analog conversion equipment (both back to back channel banks and transmultiplex equipment) were used to interface the digital switches into the analog transmission equipment. In the early 70's the digital switches were in fact "digital islands in an otherwise all analog telephone system". In this same time frame, digital fiber optics, coax and digital microwave transmission equipment were being developed and deployed for interconnecting digital switches. The conversion to an all digital switching and transmission system is now almost complete. Through the judicious use of analog to digital and digital to analog conversion equipment this evolution has taken place in a "graceful" path that did not require equipment to be discarded while it still had useful life.

A similar analog to digital evolution path is now available to the broadcast industry since all of the required digital equipment and the necessary conversion devices are currently available. The "missing links" to perform the all analog to all digital conversion (digital routers, digital fiber optic transmission equipment, and analog to digital/digital to analog conversion equipment) have recently become available. These devices, coupled with the already existing cameras, video recorders, monitors, and signal processing equipment available with either analog or digital interfaces, permit a graceful evolution path from an all analog world to an all digital world. Care must be taken in designing an "evolution path" because digital equipment has a set of design rules that are different from the set of rules necessary to make an analog system work together properly. If carefully thought out and designed properly, the transition from an analog production facility to a digital production facility can occur over a period of time without having to discard equipment that still has useful life and can be accomplished in evolutionary steps.

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SBE DAY
DEALING WITH DISASTERS

Tuesday, April 20, 1993

Moderator:
Jerry Whitaker, Technical Writer, Beaverton, Oregon

HURRICANE ANDREW: SURVIVING THE "BIG ONE"
Stephen P. Flanagan
Post-Newsweek Stations, Inc.
Miami, Florida

***LESSONS LEARNED FROM THE LOMA PRIETA EARTHQUAKE AND THE LOS ANGELES RIOTS**
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KFWB Radio
Los Angeles

***THE BAY AREA EXPERIENCE: EARTHQUAKE AND FIRE**
Roy Trumbull
KRON-TV
San Francisco, California

ROUNDTABLE
All Participants

*Paper not available at the time of publication.

HURRICANE ANDREW: SURVIVING THE "BIG ONE"

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Post-Newsweek Stations, Inc.
Miami, Florida

Hurricane Andrew a fierce hurricane caused significant property damage and death in Dade County Florida in August 1992.

The outstanding response by the National Hurricane Center and the Miami area broadcasters played significantly in keeping the death toll to a minimum.

Many lessons about disaster preparedness were learned. This paper will address several of those lessons and detail some specific thoughts for preparing a disaster plan for a broadcaster.

On Monday morning, August 24, 1992, Hurricane Andrew, the first named storm of the 1992 hurricane season, slammed into south Florida.

The eye of the storm came ashore at 4:45 AM at Homestead, Florida, in southern Dade County. At the time of impact the storm was classified as a category four storm by the National Hurricane Center using the Saffir-Simpson scale with maximum sustained winds of 145 miles per hour. Sustained winds, by definition, blow continuously for at least sixty seconds. The highest peak wind gust *recorded* was 175 miles per hour, however the National Hurricane Center's preliminary report on the storm dated September 16, 1992, suggests that even higher sustained winds and gusts were likely over parts of southern Dade County.

South Florida had not encountered a hurricane in over a decade; had not encountered a major hurricane in 27 years; and had never encountered anything of the magnitude of Hurricane Andrew. Hurricane Andrew is now ranked as one of the three worst storms to make U.S. landfall. The official death toll in Florida and Louisiana as a result of this storm was 54. While 54 deaths is a significant total, given the ferocity of the storm, this toll is low thanks to excellent storm tracking and prediction by the National Hurricane Center and the response by the south Florida news media that began on Saturday, August 22, to provide extended news coverage. By Sunday, as it became apparent that the storm was headed directly toward Miami, virtually all the radio and television stations were providing continuous news coverage and important emergency and evacuation information.

Miami began to feel the effects of the storm around 2:00 AM, first with squall lines, then slowly but steadily the winds increased reaching their maximum of 145 miles per hour around five in the morning. All told, the winds howled for more than five hours as the storm continued to move due west at a rapid pace. When the calm and the morning light arrived, southern Dade County was left demolished. Hurricane Andrew's destruction has been estimated at between 15 and 30 billion dollars, making it the single largest natural disaster in U.S. history.

Hurricanes are a concern along the East Coast or the Gulf Coast of the United States just as earthquakes are a concern in California. If you choose to live and work in one of these areas you must be prepared, both on a personal level and at your place of business. If your business is a radio or television station not only must you protect your assets, but you must be prepared to provide the news and information that the public has come to expect from you every day.

If viewers tune to you for information on a regular basis and then, when they need you most, you are not there, the long term effect on your business could be dramatic.

While Hurricane Andrew was not a surprise to south Florida when it made landfall early that Monday morning, it was classified as only a medium intensity tropical storm when most businesses closed at 5:00 PM on the previous Friday night. This storm was the first of the 1992 hurricane season, and remained a tropical storm with little potential for danger until Saturday, August 22, at 11:00 AM when it was first declared a hurricane. It then strengthened very rapidly and became a major threat within hours.

This experience brings to mind that old boy scout motto *be prepared.* More specifically if you are involved with the technical aspects of a radio or TV station, it could be more bluntly stated: *don't go home Friday night without filling the generator fuel tank.* While it may be very obvious advice ask yourself, *am I prepared right now?* Total honesty may point out several issues that are outstanding.

Rather than being lulled into a false sense of security, be a pessimist. Remember *if you haven't used it in along time, it probably won't work. If it does work initially, it will probably soon fail.*

Lesson #1 *Really test your generator.*

Test run your generator often, under anticipated loads for extended periods of time. Testing an hour a week might not uncover overheating, oil burning or a million other problems. Remember you are soon going to depend on a mechanical system that is most likely old, and not used very often, and you are going to ask it to perform flawlessly for as long as several days.

Again think like a pessimist. *Do I have oil, spare fan belts and radiator water and just as importantly, do I have a way to get them into use should the need arise?*

Lesson #2 *Buy early.*

Remember that if you need something *after* the disaster strikes, it is likely that everyone else will need it too. Therefore buy early or at the very least make concrete plans. Keep in mind that in a state of emergency even broadcasters will not be on the top of the priority list for certain items. Recognize that public safety takes precedent. Hospitals, police and fire rescue always get top priority.

Immediately after Hurricane Andrew, south Florida came to a complete stop for two days. There was no electricity in 80% of the county, therefore no businesses were open, no traffic lights worked, most phones were out of service, and cable TV, forget it.

It was nearly impossible to move around the county due to downed power lines, trees and debris. What little traffic that did move, did so at a madding crawl. Commute times of two hours were not unusual on a normal 20 minute trip.

As a recap, Lesson #1, treat your generator with respect, have a good supply of clean fuel on hand, and know that your generator is going to withstand the emergency by testing it often and for extended periods of time under realistic loads.

Lesson #2, if you need something or even think you will need something *buy it early.* Remember that during and immediately after a disaster nothing is normal.

The above lessons are very straight forward and very obvious things to plan. If I left you with only those thoughts I would be doing you a great disservice. What I would now like to cover are some of the things that might not be quite so obvious, but are just as important and will certainly make a difference in your effectiveness if there is a disaster.

One of the most overlooked items at most radio and television stations is a meaningful disaster plan. Disaster plans are something that everyone agrees should be done but somehow few ever get around to putting anything on paper. Therefore if I can impart only one lesson from the Hurricane Andrew experience it should be **MAKE A DISASTER PLAN.**

Lesson #3 ***Make a Disaster Plan***

A disaster plan should be in one document a summary of your station's vulnerabilities and some potential remedies. The plan must take into account your particular situation, it must be realistic and it must *distribute* responsibility.

Hurricane Andrew taught us that when an unusual event occurs, non-technical people tend to look to those who know how. Remember the show Gilligan's Island. After a short time everyone depended on the Professor. Therefore in your planning remember that not every responsibility should fall on the Engineering Department. At WPLG we had setup teams from the Business and the Programming Departments to deal with the food and water gathering and the transportation issues. We also reassigned members of the Sales Department to help both News and Engineering.

Your disaster plan must be created as a group, not written by some curmudgeon in the Engineering or News department and then filed away in a cabinet only to be dusted off and read for the first time when the big event arrives. It must be a living document that all of your key employees had input into and more importantly, believe in. Fine tuning a disaster plan the day before the event is fine, creating it is not.

One thing about hurricanes, thanks to modern tracking and reporting technology, is that they are not a surprise; station fires, tornadoes, earthquakes and riots are. Therefore the disaster plan must be current. Review it at least annually, update it accordingly and share it. Recall that staff turnover, at least in some departments, is significant.

Lesson #4 ***Keep in mind that your employees are human.***

For the most part employees have families, friends, pets and property that they will feel need protection also. *Most employees will be torn between the needs of their personal situation and their responsibility to the station.*

Since it is impossible to predict your programming or news requirements in advance, set up a disaster schedule that gives employees at least an outline of what you will expect from them. For example: When your plan is put into effect, Operating Technicians will be divided into two teams. Each team will cover a 12 hour shift, seven days a week. This simple step will alleviate a tremendous amount of confusion especially in those first few critical hours.

Don't burn out your entire staff just before, during or immediately after the event. Take some time to consider the magnitude of the disaster, then gauge your immediate and future needs accordingly. When employees are tired they are not at their best and this may be at a time when you need their efforts and skills the most.

Lesson #5 ***Be prepared for differing and sometimes surprising actions from your staff.***

Everyone will be effected to one degree or another by the situation depending on their normal emotional make-up, their particular immediate and personal situation and their level of responsibility. Be sensitive to those issues, be prepared to make changes, to counsel and even send home those employees whose personal situation warrants.

I don't mean lesson #5 as all negative. I had several medium level employees whose positive response to the situation was extraordinary. They were everywhere, doing everything: a real asset. When everyone is under great stress, you may be surprised, sometimes pleasantly.

Lesson #6 ***Don't be afraid to ask for help.***

Immediately after the storm many out-of-town broadcasters and manufacturers called to lend a hand with both equipment and personnel. If you need it, use it. Remember that you will be tired, your staff will be tired and your work load has just increased dramatically. Don't let the enormity of the situation overwhelm you. Divide the tasks accordingly and realistically.

Lesson #7 ***Being "On the Air" and being "On the Air Competitively" are not the same.***

In the 1990s, viewers have come to expect a lot from broadcasters. If a story happens from around the world your viewers expect to see pictures and expect you to put the story in the proper perspective. When the story happens in their town they expect even more. In a competitive situation you must be able to deliver at least as well as the other guy. Station rating can be made or broken during a major event.

Lesson #8 ***Be prepared for an invasion of media***

Not only will your own News and Programming departments want you to deliver the world, but broadcasters from everywhere will descend upon you and they'll all need something. Best advice: be fair, honest and keep their needs in the proper perspective. Long term relationships are very important and must be considered, but outsiders must be sensitive to your situation.

Lesson #9 ***Be prepared for the unexpected.***

A quick synopsis of the major equipment and facility damage to Miami television stations may best illustrate this point.

WCIX, the CBS owned and operated station on Channel 6:

Their main 1800 foot tower collapsed. The station was off the air for seven days except for a low power repeater site in Broward County. Clearly their engineering efforts were focused on locating transmitters, transmission line, an antenna and a temporary tower site to get back on the air.

WPLG, the ABC affiliate Channel 10 owned by Post-Newsweek:

The studio microwave tower collapsed, which ruined 2 STL links, 6 TSL links, a link to ABC network, a link from ABC, a link to the National Hurricane Center and the link to the Dade County Emergency Operation Center.

We were off the air for 3 hours. Back on without any ENG live capability for 24 hours. Our efforts were centered on reestablishing microwave paths by locating dish antennas and constructing temporary mounts on rooftops. Like **WCIX** we needed tower crews for some repairs.

WTVJ, the NBC owned and operated station on Channel 4:

They remained on air the throughout the storm. Damage was centered on dislocated microwave dishes. Just like everyone else, they needed the assistance of tower crews for repairs.

WSVN, Channel 7 the Fox Affiliate News Station:

They evacuated their main studio along the causeway to Miami Beach fearing a storm surge. They broadcast from their transmitter site for several hours with both news and weather personnel. Immediate damage was minimal, but their main antenna mast bent in the high winds and will require replacement. They remained on the air throughout the storm.

As one can see, the damage to the different stations varied depending on location, equipment, facility design and luck.

It will be impossible when creating your disaster plan to accurately identify every point of vulnerability. Therefore try to think in broad terms when detailing your potential responses.

Conclusions

The best response to a disaster is to plan. The best way to plan is to create a disaster plan that is detailed to a practical level, takes into account the potential vulnerabilities of your particular situation, distributes the responsibilities interdepartmentally and is shared and agreed to by the key managers and employees of your station. It will be impossible to out-think every situation, but some planning will give you a real advantage.

SBE DAY
EBS SUMMIT CONFERENCE

Moderator:
Jerry Whitaker, Technical Writer, Beaverton, Oregon

EBS: WHERE WE GO FROM HERE?
Dane Ericksen, P.E.
Hammett & Edison, Inc.
Consulting Engineers
San Francisco, California

***WHERE THE REGULATORS STAND ON EBS**
Richard Smith
Federal Communications Commission
Washington, District of Columbia

UPGRADING THE EMERGENCY BROADCAST SYSTEM
Frederick M. Baumgartner
TFT, Inc.
Santa Clara, California

EBS ROUNDTABLE
All Participants

*Paper not available at the time of publication.

EBS: WHERE WE GO FROM HERE?

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Abstract

The FCC has proposed major revisions to the Emergency Broadcast System. This paper reviews those proposals, and summarizes the future of EBS as envisioned in comments filed by the Society of Broadcast Engineers, Inc. (SBE).

INTRODUCTION

In October, 1992, the Federal Communications Commission (FCC) issued a combined Notice of Proposed Rule Making and Further Notice of Proposed Rule Making ("Notice") continuing its broad-based inquiry, initiated in 1991, into possible improvements to the Emergency Broadcast System (EBS). The Notice represents the first major overhaul of the Emergency Broadcast System since 1975, when the system was modified to use the two-tone Attention Signal and the Federal Emergency Management Agency (FEMA) and the National Weather Service (NWS) were brought into the system.

Major changes to the Emergency Broadcast System are now proposed. The system will probably be re-named. The existing two-tone Attention Signal will almost assuredly be shortened and is proposed to be abandoned as an *inter-station* signaling technique by 1995. A new alerting system with multiple inputs and utilizing at least two different signaling protocols will most likely be substituted, and it is proposed to become mandatory by 1994. Cable television systems would be brought into the system.

SUMMARY OF PRESENT EMERGENCY BROADCAST SYSTEM AND ITS PROBLEMS

A review of the existing Emergency Broadcast System and several of its widely perceived problems will be helpful before discussing the proposed new and improved system.

The present Emergency Broadcast System is an inband audio signaling system. It uses an alerting signal, called the Attention Signal, comprised of 853 Hertz and 960 Hertz tones, for this purpose. The Attention Signal duration is now 20-25 seconds, although a shortened interval of 8-25 seconds has been proposed and will likely be adopted. All broadcast stations, except 10-watt noncommercial Class D FM stations, must now maintain equipment capable of generating and detecting the Attention Signal. The current system relies on the Attention Signal to alert duty operators at broadcast stations that an emergency message may be forthcoming. It is a manual system, requiring the proper intervention of a human operator at each point in the chain of relaying broadcast stations in order for emergency programming to ultimately reach the public in a timely manner.

The shortcomings of the current Emergency Broadcast System include the following:

• It suffers from a "cry wolf" syndrome. Because the present EBS system mandates weekly over-the-air tests that must include the Attention Signal, the public has become de-sensitized to that signal, and accordingly an actual, non-drill Attention Signal may be ignored until it is too late.

• The Attention Signal tends to be a "tune-out" factor, a problem made worse by the wide availability of digitally tuned receivers. Push one button and your two tone headache is gone.

• The current EBS has only a single alerting signal; that same signal is used for national level alerts, state level alerts, local level alerts, and tests. An improved EBS needs to include codes to indicate the seriousness of the alert.

• The current EBS tends to include too broad an area when it is activated. For example, in the case of severe weather situations, the storm of interest may be located on the eastern edge of a station's coverage area. Activating the

EBS tones each time a tornado or other severe weather is observed as the storm tracks across the large coverage area of a Class I Clear Channel AM station or a Class C FM station can make the cry wolf and tune-out problems even worse.

• The current EBS is sometimes too slow for time-critical warnings such as tornado alerts. Some mid-west stations already use a tornado "sounder", give the pertinent information about the tornado, then follow it with the EBS message.

• The current EBS only works well for emergencies that are conducive to dissemination of information from an informed, centralized authority: for example, a NWS tornado alert or a tsunami alert. The system tends to perform poorly when an emergency that is not conducive to an informed governmental source occurs: for example, immediately after a major earthquake, or in the event of widespread civil unrest. The 1989 Loma Prieta earthquake near San Francisco and the 1992 Rodney King verdict riots in Los Angeles are good examples of situations not well suited to the current EBS structure.

• The current EBS suffers from an "all or nothing", or a "decision indecision" problem. Government officials often delay activation until the emergency is patently obvious, to avoid being accused of an unwarranted activation of the EBS system

• The current EBS is only conducive to accepting information from a single source. Emergency information often comes from many different sources. An effective emergency alerting system needs to be able to monitor and accept input from many sources.

• The current EBS is not routinely used by operators in their day-to-day duties. Rather, the system is only used during weekly tests and actual alerts, making it prone to operator error when an actual emergency does occur. Systems and procedures that are used and practiced every day are the ones *most likely* to be remembered during times of great stress.

• The current EBS is prone to having real alerts missed because operators may reset their EBS receiver before they know whether the transmission is a real alert or just a test.

• The current EBS does not include cable television systems or Multichannel Multipoint Distribution System (MMDS) stations (so called "wireless cable" systems).

• Finally, and not least, the current EBS has turned into a financial threat to broadcasters because of the possibility of huge fines for even minor infractions of the Byzantine EBS rules. For example, the standard forfeiture for EBS equipment not installed or operational is $12,500, according to an August 1, 1991, FCC Policy Statement, "In the Matter of Standards for Assessing Forfeitures".

Proposed New System

The FCC proposes that the minimum duration of the current, two-tone Attention Signal be shortened from its present 20 seconds to 8 seconds, and that use of the Attention Signal as an inter-station signaling method be discontinued after July 1, 1995. The maximum Attention Signal duration would remain unchanged at 25 seconds, and the Attention Signal would still be used for alerting the public. For inter-station signaling, a replacement alerting scheme offering varying levels of automatic alerting has been proposed.

Automated Alerting Systems

Four automatic alerting systems have been developed: the National Weather Service's WRSAME, Colorado's ICEBS, Sage Alerting System's RDS, and California's EDIS. These systems can be summarized as follows:

WRSAME The WRSAME system, or Weather Radio Specific Area Message Encoder system, uses a network of 380 transmitters across the county. These federally operated stations transmit signals on three frequencies in the 162 MHz band and are intended for direct reception by the public, state and local governmental agencies, mariners, pilots, and, of course, broadcast stations. Over 90% of the U.S. population is served by these NWS stations. The system is in some ways similar to the present EBS, in that it can send an inband alerting message, in addition to its normal aural weather reports. However, unlike the EBS Attention Signal, the WRSAME alerting signal is a digital signal. The WRSAME alerting signal uses a binary frequency shift keying modulation system with a 520 baud signaling rate. Zeros are encoded at 1,562 Hertz and ones are encoded at 2,083 Hertz. Further, the WRSAME alerting signal identifies the type of emergency and the affected counties. Therefore consumer receivers built to receive and decode the 162 MHz NWS transmissions give the user the option of selecting only certain types of emergencies or specific counties for alarm activation. This selective coding feature would also allow a broadcaster to select the types of emergencies and areas that would activate automatic re-transmission over their broadcast station.

ICEBS The ICEBS, or Improved Colorado Emergency Broadcast System, was jointly developed by the Colorado Broadcasters Association, the Colorado chapters of the SBE, FEMA, the Colorado Division of Disaster Emergency Services, and the FCC's Denver field office. ICEBS is an inband system that replaces normal programming material with modified* dual-tone multi-frequency (DTMF) tones to generate codes identifying the geographic areas affected by the alert. ICEBS lacks the WRSAME coding for the type of emergency, but offers more-detailed coding for the affected areas. ICEBS uses the same scheme as WRSAME for conveying how imminent the emergency is. ICEBS therefore allows for automatic activation of EBS and the plan recommends that all stations implement such automatic activation for the "most imminent" emergency code.

RDS Sage Alerting System's RDS, or Radio Data Service system, uses a subcarrier at 57 kHz to relay emergency alerts. As such, RDS is presently only available to FM broadcast stations, a limitation. The RDS system can identify the type of emergency involved so that receivers can be selectively activated. Car radio receivers can even be programmed to continue to monitor for emergency alerts when the radio is turned off or is receiving audio from a tape or compact disk. The RDS system also has self-testing capabilities so that system failures can be readily detected.

RDS differs from WRSAME and ICEBS in that no interruption of the inband audio must occur in order to relay the emergency message codes (digital coding for WRSAME, DTMF coding for ICEBS). Thus, RDS does not disrupt normal programming and therefore does not limit how such a system could be used by broadcast stations in non-emergency situations (including system tests).

EDIS California's EDIS, or Emergency Digital Information Service system, can be thought of as a government access newswire. It is a packet radio system that uses Local Government or Special Emergency land mobile radio frequencies to disseminate emergency information to local governmental agencies and to participating broadcast stations. EDIS employs an ASCII data delivery system using RS-232 format at 1200 baud. Standard American Newspaper Publishers Association (ANPA) header coding is used so it can be easily folded into newsroom computer systems.

EDIS is not used for alerting the public directly. It is therefore a non-inband alerting system. A potential drawback of the EDIS system is its need for additional frequency spectrum, although if an existing Public Safety or Special Emergency channel is available and can be utilized this disadvantage disappears.

* Modified DTMF tones, with frequencies shifted upward by 12%, are used to differentiate the tones from those used by the switched telephone network, to decrease the likelihood of a false activation. The modified tones can be obtained by substituting a 4 MHz crystal for the 3.58 MHz crystal commonly used by DTMF-generating and DTMF-decoding integrated circuits.

Levels of Automatic Alerting

All of the new systems would allow automatic alerting. The Notice defined three levels of automatic alerting: Type I, Type II, and Type III. Those levels can be described as follows:

Type I Type I activation of emergency programming is fully automatic. The duty operator would not have a veto option. Type I activation has the advantage of speed in relaying an emergency message to listeners, but runs the risk of false alarms, due to either inadvertent or malicious errors.

Type II Type II activation of emergency messages allows the duty operator to suppress, or veto, the pass through of an incoming emergency message. This would require a buffer between the time the message is received and the time the automated system seizes the transmitter; it therefore adds an inherent time delay, and requires constant vigilance by the duty operator if full oversight is to be maintained.

Type III Type III activation of emergency messages would require an acknowledgment by the duty operator before the equipment would seize the transmitter. Type III activation provides an improved level of protection against false alerts, but also increases the risk that a bona fide emergency message may be delayed, or not relayed at all, due to operator error.

The WRSAME and ICEBS systems give broadcasters the choice of all three levels of activation, while the RDS system focus on Type II activations. The Notice proposes that Type III activations continue to be the default mode for most broadcast stations, and continuous monitoring by a duty operator would remain mandatory for Common Program Control Stations (CPCS), Primary Entry Point (PEP) stations, Originating Primary Relay (OPRELAY) stations, and Relay (RELAY) stations. The Notice indicates that operator oversight will still be necessary for such stations, as they will continue to provide the first level of emergency warning to their regions, states, and communities. However, the Notice proposes that CPCS, PEP, OPRELAY, and RELAY stations be given an option to use Type II or Type I alerting for area-specific emergency information consistent with their state and local EBS plans.

New Emergency Alerting Device

A new piece of electronic equipment, called an "emergency alerting device", or simply "Device", is proposed. This Device would initially augment, and then replace, existing EBS encoders and decoders. The Device would allow multiple input and output parameters and would serve as a link in the programming chain. The Notice proposes that the Device be capable of simultaneously monitoring two off-air broadcast signals, one AM, one FM. The Notice also asks whether the Device should be capable of monitoring VHF or UHF land mobile frequencies. The Device would also have to be capable of generating the WRSAME and RDS codes.

The Device would have to incorporate a self-testing feature to ensure the proper working of decoders and encoders, the presence of RF carriers at the appropriate inputs, and visual displays to indicate operational readiness.

The Notice proposes certain optional features for the Device. These include automatic recording and/or hard-copy printouts; automatic dialing or paging of critical station personnel, in the event of an alert; and a data port for receiving digital signals.

The Notice envisions that three versions of the Device would be needed: one for AM and TV stations, one for FM stations, and one for cable television systems. The Device characteristics would be as follows:

AM/TV Device Able to detect the digital WRSAME codes broadcast by a NWS station, or relayed by a broadcast station. Able to detect RDS codes broadcast by FM stations. Upon receipt of these codes, and consistent with the level of automation selected, the Device would override the station's regular programming, generate a shortened EBS Attention Signal, pass through the emergency message, and generate an end of message code.

FM Device Able to detect the WRSAME and RDS codes, capable of relaying the WRSAME codes on the station's inband audio signal, and additionally capable of generating the RDS codes on the station's 57 kHz RDS subcarrier. Upon receipt of these codes, and consistent with the level of automation selected, the Device would again override the station's regular programming, generate a shortened EBS Attention Signal, pass through the emergency message, and generate an end of message code.

Cable System Device Able to detect the digital WRSAME codes broadcast by a NWS station, or relayed by a broadcast station. Able to detect RDS codes broadcast by FM stations. Able to detect DTMF signals used by the switched telephone network. Upon receipt of WRSAME or RDS codes, or the appropriate combination of DTMF signals, and consistent with the level of automation selected, the Device would interrupt audio programming on all non-broadcast cable channels. The Device would override the cable system's non-broadcast programming, generate a shortened EBS Attention Signal, pass through the emergency message, and generate an end of message code.

The Notice estimates the cost of the Device for all broadcast and cable systems would be between 50 million and 60 million dollars. Since there are approximately 11,300 radio stations, 1,500 TV stations, and 11,000 cable systems in the United States, this calculates to approximately $2,300 per Device.

As with many government cost estimates, this is probably too low. A more realistic cost estimate is $3,000 per device, or about $70 million for the broadcasting and cable industries combined.

Remote Control

An automated alerting system does not negate the need for remote control capability when Type II or Type III levels of automation are chosen. Therefore the Device needs to capable of remote control operation. Further, the Device needs to be compatible with newer remote control technologies, such as dial-up remote control systems and satellite-based remote control services. A compatible remote control capability will encourage greater participation in a revamped emergency alerting system. For these reasons the Notice concludes that the Device must be compatible with modern-day remote control systems.

Monthly Instead of Weekly Attention Signal Tests

To minimize the cry-wolf and tune-out problems, the Notice proposes only transmitting over-the-air tests using the Attention Signal once per month, instead of weekly, as is now required. Further, this monthly test would be simultaneously transmitted by all stations and cable systems in an EBS operational area or state. Scheduling of the monthly tests would be done by the CPCS-1 station for the operational area or by the OPRELAY stations for state-wide areas.

Weekly tests by all broadcast stations would still be required, but would be "silent tests" not resulting in transmission of the Attention Signal or an audio/video test script. Weekly silent tests would require transmission of the WRSAME and RDS codes.

Revisions to the EBS Test Script

The Notice indicated there were relatively few comments received regarding revisions to the test script ("This is a test. This stations is conducting a test of..."). Nevertheless, the Notice indicated that the Commission would be open to giving broadcasters more latitude in what goes into the test script, possibly allowing Public Service Announcements (PSA's) to be broadcast in conjunction with the test script.

Prohibition Against False Use of the Attention Signal

For those few operators who need things explained to them the hard way, the Notice proposes to add a section to the EBS Rules prohibiting the false or deceptive use of the Attention Signal or the proposed WRSAME and RDS codes. It seems incredible to this author that any person would need this explained, but sadly certain incidents have occurred that demonstrate a need for such a rule.

SBE COMMENTS

In further comments filed on January 15, 1993, by the SBE, the SBE envisions a system that would provide for the transmission of voice and data messages from state and local government authorities, and by satellite from federal authorities, for the dissemination of emergency information to the public. The SBE believes that the key to a reliable emergency alerting system is to give broadcasters a tool that is up and running (and used) virtually all of the time. The system should therefore integrate emergency information into the regular framework of a station's information dissemination. It should permit emergency notifications to be based on information received directly from the source of the emergency alert, on a federal, state, or local basis. It should allow assessment of the local impact of the information, and should not allow the listening public to be "numbed" by repeated EBS alert tones. It should encourage media personnel to become a partner with the sources of emergency information.

The Emergency News Network

The SBE proposes re-naming EBS to the Emergency News Network, or ENN.

The ENN would be used routinely by station personnel, so when life threatening natural disasters or other events warranting activation of the ENN system occur and actual emergency traffic starts flowing, station personnel are more likely to know what do to. The ENN would allow inputs from multiple sources, and not just a "from the top down" relaying system. The ENN system envisioned by the SBE would recognize that most government personnel go home at night and on weekends, that the news media doesn't, and that government gets much of its initial information during an emergency situation by listening to the news media.

This approach would make the ENN an "RMS function" alerting system as opposed to the "step function" nature of the current EBS. ENN would break the "all or nothing" nature of EBS and allow broadcasters to provide earlier information on a developing emergency. It would also help mitigate the "decision indecision" problem by public officials that sometimes hampers the present EBS.

Stations should, upon receipt of the information, be able to select their appropriate level of participation within the ENN structure and determine, through hardware and software configurations, whether automatic or manual override of regular programming is justified in a particular case. The ENN system should be used regularly for transmission of status information not related to emergency alerts, such as bulletins, supplementary information, and the like.

Replacement of the regular EBS tests with ENN promotional messages, to encourage each station's participation in ENN, and to foster familiarity with ENN procedures in the event of an emergency, should be permitted. It could also result in a more realistic test text, one that would not make promises that can't always be kept ("...have developed this system to keep you informed in the event of an emergency").

The Commission should attempt to create a more positive view of EBS generally, and, rather than imposing forfeitures on broadcasters for violation of technical details of the currently complex EBS rules, unless repeated or egregious, the new ENN rules should be aimed at integrating emergency communications information into the routine functioning of the broadcast media.

The ENN Device

The ENN Device should be configured so as to be operable, and its status readily determined, by non-technical operating personnel at broadcast stations and cable television systems. The status of its functions should be readily determinable by computer interface or by visual indication, and its status remoteable in that same fashion. In the event of malfunction, the device should notify the user of any condition in the system that could indicate faults, including the nature of the problem. A status report should be automatically logged periodically, to provide a record for the licensee of system operation. This is especially important for heavily automated stations that carry satellite programming and for stations operating by control services such as the National Supervisory Network. Diagnostic information should alert the non-technical operator to appropriate remedies.

The ENN Device should be capable of accepting EDIS packet data codes, in addition to WRSAME and RDS codes, and DTMF tones. Digital as well as voice information should be sent to any stations that want it, by way of whatever network is available. This avoids the fragility of the current CPCS mold that can fall apart if a few key stations fail to activate or are off the air due to the emergency.

An ENN Device with EDIS capability would provide a way for TV stations to broadcast open captioning during emergencies. This would be especially useful to alert hearing-impaired citizens who are often left without any means of emergency warnings if their Telecommunications Device for the Deaf (TDD) equipped telephones fail.

The ENN should not be in a "daisy chain" or "domino" configuration, but rather should be configured in a "web" format, as is done under ICEBS. This would necessitate a monitoring arrangement for relaying alert information, so that failed primary stations would be known to the monitoring station, and it would be apparent that alternative alerting information would be necessary.

SBE Views on RDS

The SBE sees RDS as a helpful tool, but believes it must be viewed as only one leg of a WRSAME-ICEBS-RDS-EDIS quartet rather than as a stand-alone alerting system. As a system currently available only to FM stations, RDS places AM stations at a competitive disadvantage. Further, FM stations tend to be clustered together, making them vulnerable to simultaneous failure due to natural disaster or terrorist action. In contrast, AM stations tend to be spread out at multiple sites, making it unlikely that a large number of AM stations in a given area could be simultaneously taken off the air. Especially for all news format AM stations, which typically have spent years "hardening" and making their facilities redundant, it is those stations which are most likely to remain on the air in the event of a major natural disaster.

Time Frame

The SBE believes that the proposed July 1, 1994, implementation deadline for a new alerting Device is not sufficient for the development, testing, equipment authorization, marketing, and inclusion in the budgets of broadcast stations and cable television systems. A more reasonable timetable would be July 1, 1995.

Housekeeping Issues

The SBE believes that there are several "housekeeping" issues raised in the initial SBE comments to the EBS docket that were apparently overlooked in the Notice. The first housekeeping issue is for the Commission to resolve the current uncertainty that exists concerning the proper modulation level for the Attention Signal.

Section 73.906 of the FCC Rules states that the 853 Hertz and 960 Hertz tones comprising the Attention Signal shall be calibrated separately to modulate the transmitter at no less than 40%, and that the two tones shall have modulation levels within 1 decibel of each other.

Many stations employ a downstream limiter or other processing device that individually will pass each of the Attention Signal tones at 40% modulation but in the presence of both tones causes a composite modulation level of much less than 80%. Although the impact of an undermodulated Attention Signal will be reduced once alternative inter-station relaying methods are implemented, there are still many EBS decoder/receivers located in businesses and homes in the tornado belt and near nuclear power plants that would continue to use the Attention Signal as their primary alerting method. These receivers will not unsquelch if a seriously undermodulated Attention Signal is transmitted.

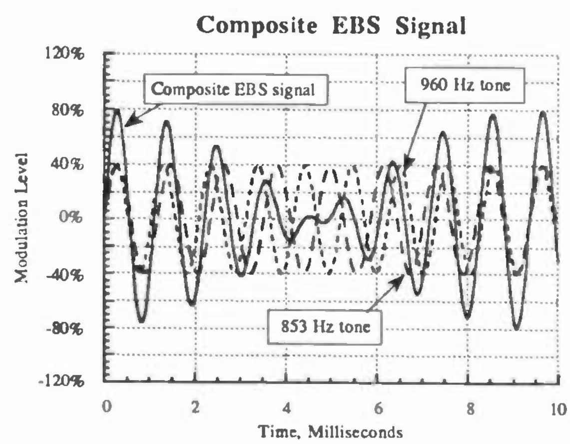

Figure 1. Composite Attention Signal showing relationship between the individual tones and the composite signal.

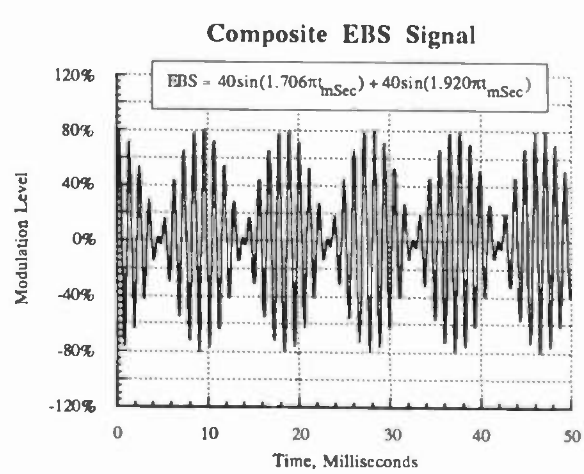

Figure 2. Composite Attention Signal showing the number of modulation peaks with time.

Figures 1 and 2 show the relationship between the 853 Hertz and 960 Hertz tones and the composite Attention

Signal. Even for a shortened Attention Signal duration of 8 seconds, there will be at approximately 850 occurrences where the two tones are in phase and a peak modulation of 80% results if each individual tone is simultaneously modulating the transmitter to 40%. Thus, the peak flasher of a properly operating pre-1983 type approved modulation monitor should easily detect the peak modulation of the Attention Signal.

The SBE comments therefore again urged that the Commission clarify whether a downstream limiter or other processing device that individually passes each of the Attention Signal tones at 40% modulation but in the presence of both tones causes a composite modulation level of much less than the algebraic sum of the two tones comply with the requirement to "separately calibrate" the modulation level. Or, in other words, is a downstream limiter or processing device that in effect destroys the single-tone calibration acceptable?

A related housekeeping issue raised by the extension of the EBS/ENN system to cable television systems is that of ensuring proper modulation levels for the Attention Signal, or other emergency messages transmitted by cable television systems. There are currently no deviation standards for the aural portion of television signals carried by cable television systems, or for the modulation level of FM broadcast station signals carried by cable television systems. This was not a problem 20 years ago, when virtually all cable signals were heterodyne processed; a heterodyne processor is generally incapable of affecting the modulation level of the signal it processes. However, it is now common for cable systems to re-modulate a large majority, and often all, of the signals that they carry. Mis-adjusted cable headend modulators have the ability to undermodulate the aural signal. Adoption of a cable television deviation standard defining ±25 kHz frequency deviation as 100% modulation for TV aural signals, and ±75 kHz deviation as 100% modulation for FM signals, would ensure that EBS/ENN alerts sent to cable television subscribers are not thwarted by improper modulation levels.

As a final housekeeping item, the Commission should ensure that open-captioned emergency messages transmitted by television stations for hearing impaired viewers do not conflict with open-captioned emergency messages transmitted by cable systems, especially where an all-channel emergency override open-captioned message is used. Perhaps different screen areas could be assigned to TV stations and to cable systems, to make sure that one on-screen emergency message crawl does not overlap an emergency message crawl from another source.

ACKNOWLEDGMENTS

The author would like to thank Mr. Richard Rudman, Engineering Manager of Station KFWB, Los Angeles, Mr. Kenneth Brown of Capital Cities/ABC Inc. in New York, and Christopher Imlay, Esq., of Booth, Freret & Imlay in Washington, D.C. for their assistance in preparing this paper. All three gentlemen serve on the SBE FCC Liaison Committee, which is chaired by the author.

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UPGRADING THE EMERGENCY BROADCAST SYSTEM

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ABSTRACT

A new EBS system must merge current technologies to provide economy, flexibility, and efficiency for broadcasters to disseminate emergency information to the public. TFT is introducing new equipment called "Emergency Information System Manager (EIS Manager)", which combines the web concept of ICEBS with the robustness of the WRSAME data stream. The EIS Manager features scanning capabilities from multiple receivers or other audio sources; interface with the 2-tone Attention Signal and with RBDS; storage and regeneration of WRSAME messages; I/O ports for interface to peripherals including computers, printers, crawl character generators, and CRT terminals; telephone automatic dialers; telephone auto-answer sets; analog and digital recorders; and external alarms and controls. It also meets the performance and cost objectives proposed by the FCC and is adaptable for cable systems by the addition of optional firmware.

The current Emergency Broadcast System (EBS) was installed in 1975. At that time, signaling technology was elementary and expensive. The now common and inexpensive microprocessor was in it's infancy.

The current EBS system focuses on a national nuclear crisis. Today, the early detection of tornados, and the risks of high-tech emergencies (nuclear plants, military bases, chemical spills, infrastructure failures), have out paced the EBSs ability to bring timely information to the public.

The proliferation of Cable TV and soon, other mass media channels, has changed the method of delivering information to the public. A number of recent emergencies have illustrated how important it is to improve the EBS.

In this paper, I will outline the basic EIS protocol. The actual hardware to implement the system is built around a microprocessor that "scans" many audio and data sources for EIS, or other emergency information. The device can save voice messages for review or rebroadcast, as well as automatically and nearly instantly rebroadcast messages as desired. The manner in which a message is handled is predetermined by the operator, and decisions are stored in memory, and recalled when emergency messages are received.

The hardware is inexpensive and universal. Software modules permit the generation of "crawls" for TV and Cable, as well as providing data to other text services, printers, newsroom computers, and the like.

At this writing (January 1993), the EIS protocol is a working document, and will likely be expanded and contain error correction in future versions. While these are easy to accomplish, the compatibility with existing WRSAME, and other signaling codes is a concern.

EIS UPGRADES EBS

The EIS (Emergency Information System) is designed to quickly carry warning and emergency information to the public.

The EIS serves, and extends existing alerting technologies. It serves the needs of national, state, local, weather, and other organizations who have information necessary to save lives and property.

The EIS utilizes all electronic media; Broadcast TV and Radio, Cable delivered services, Satellite services, Data services, NOAA Weather radio, and other government and private warning services, and other appropriate wired and unwired communications channels.

The EIS provides both data and voice communications, on in-band and out-of-band communications channels.

The EIS is capable of activating "quieted" receivers, and incorporates shortened EBS type two-tone activation signaling and NOAA single-tone activation signaling where appropriate.

The EIS permits the Broadcaster the entire range of participation from automated operation, to verify before release, or hold for retransmission.

The EIS permits targeted activations, crosses boundaries and coverage areas well, and includes a short message description at the beginning of each activation.

The EIS is compatible with WRSAME, EBS, RBDS, Teletext, closed caption, over the air and cable system warning/call devices, and other existing and planned warning and communications technologies.

The EIS is extremely robust, and capable of functioning even when a substantial portion of the communication system has failed.

The EIS reduces the opportunity for human error, and delays that result from indecision.

BACKGROUND; WHERE EIS CAME FROM

The EIS is a combination of technologies previously offered as part of three major proposed EBS replacement or supplementary systems. The signaling protocol of the WR-SAME (Weather Radio Specific Area Message Encoder) is preserved in it's entirety. Likewise, the "WEB" architecture and basic controls and interfaces of the EAS (Emergency Alerting System; sometimes called the ICEBS, or Improved Colorado Emergency Broadcast System). Further, it includes the data capacity of the EDIS (Emergency Data Information System) that was preserved inside the EAS. In addition, the audible warning tones used by currently installed EBS and NOAA weather radios is also preserved and fully supported, in both manual and automatic modes.

OVERVIEW; WHAT IS IN AN EIS MESSAGE

The EIS protocol consists of a header containing location and nature of alert "headline" information. This information is decoded by a receiver system that scans between several information sources (radio transmitters, wire line, satellite, etc.). The decoded information is processed by the receiver to determine what action, if any, is called for. The information is logged by the receiver and stored for retrieval either as text or voice for broadcast, rebroadcast, test, or informational uses.

SYSTEM SPECIFICS

I. ENCODERS: Encoders can be portable, or fixed. Each unit has an exclusive "serial number" alpha-character identification which is transmitted as part of the header. Encoders can be built on a computer platform for the most complex, wide area devices, a mid-sized rack mounted unit with direct button access of commands, or a small unit with preprogrammed alert sequences for mobile applications.

II. DECODERS: Decoders "scan" or monitor multiple (two or more) sources looking for the EIS preamble and data. Decoders can range between the unsophisticated device that "unmutes" when the EIS preamble is received, to decoders that are capable of deciding when to automatically rebroadcast EIS information with EIS or NWS tones, or save and hold messages for retransmission. All units are capable of displaying short text messages, and full service units are capable of printing text, or preparing text for rebroadcast via data communications systems such as Teletext, TV "crawls," closed caption, RBDS (Radio Data System), or other data services.

Decoders that are required to pass the EIS message, do such by decoding the original EIS header, data and EOM (end of message), then regenerating the FSK (Frequency Shift Keying) EIS signal.

Consumer units can employ a number of means to activate for an alert. Detecting the EBS tones, the NOAA tone, the EIS preamble, or a properly encoded RDS/MBS (where available) message will allow citizens to become quickly informed of emergency situations. Further, the EIS decoders can drive proprietary alerting equipment, such as proposed in the cable industry.

III. COMMUNICATIONS PATHS: Virtually any communications path can be used to relay EIS messages. The following is a partial list:

A. Over-the-air broadcast. It is anticipated that the existing EBS, and proposed PEP (Primary Entry Point) networks will remain functional, using the EIS signaling protocol, supplemented by other communications paths.

B. NOAA weather radio. It is anticipated that the NOAA weather radio stations will serve as a primary distribution point (where available) to broadcasters, cable and other systems, using the EIS signaling protocol. They will also continue to serve as a public distribution channel for weather radios owned by the public.

C. National Emergency Frequency. At such time that a National Emergency Frequency is established, this too can serve as a distribution channel.

D. Wireline systems. There is already in existence a number of wireline services, either as "dial-up," "ring-down," or "announce" lines. These can carry the EIS messages to subscribers for broadcast distribution.

E. Public safety radio. Channels used normally for police, sheriff, fire, etc.. can serve as distribution channels for EIS messages.

F. Network (Satellite) channels. These channels are available on a national and regional level to most broadcasters and cable systems. Currently there is no signaling scheme allowing their use as emergency information channels. EIS messages can be sent on these channels.

G. Nontraditional paths. In some cases, paths such as Amateur radio services, private radio common carriers (RCCs), and the like may prove to be desirable means of relaying EIS messages. While the EIS messages can easily be sent via these links, it is assumed that they will be secured from public access through the use of encoding or other security technologies now commonly available.

IV. THE EIS "WEB" NETWORK. The EIS depends on a "web" network for reliability, and full access to broadcast, cable and other means of distributing the EIS message directly to the public. The web network allows overlapping media service areas, and public safety information to be distributed outside of the affected community to broadcasters with wide area coverage. Thus, the web network allows media serving several states or communities to carry EIS messages to their audiences.

The web network requires that decoders be capable of monitoring two or more channels, and suggests that emergency information sources should employ two or more paths to send their EIS messages to the broadcasters.

The actual layout of the web, assuring that each media outlet has access to the EIS information for their area, and that redundancy and shortest path routing has been employed wherever possible, would become part of the state EIS plan. It is assumed that the foundation of the EIS plan would be the rejuvenated EBS tree and branch network plan.

V. THE EIS MESSAGE STRUCTURE. The EIS message is composed of the following blocks of information (See Figure 1):

A. The HEADER. This carries

information necessary to route the message, and determine how that message should be processed.

B. The ALERT TONES. This is optional in that only certain channels will require special attention signals. Several signaling schemes have been installed and will remain in use for the foreseeable future. In the case of broadcasters it is the EBS tones, and in the case of NOAA, the single alert tone must still be included in order activate existing receivers. For that reason, an eight second period is built into the EIS protocol for transmission of these tones, if requested, and if authorized by the broadcaster, either automatically or manually.

C. The VOICE ANNOUNCEMENT. This is optional, in that data may be sent in leu the voice message. For most purposes, voice is a speedy, and appropriate mode of emergency communications. Under EIS, the voice portion of the message can be up to one-and-one- half minutes in length.

D. The DATA MESSAGE. This is also optional, in that some sources will not desire to or, be unable to generate a data message. The data message may also be distributed some time other than the voice message. The data message is generally used to create TV graphics (crawls), feed EDIS type systems, RBDS, closed captioning and other text based services.

E. The END OF MESSAGE (EOM). The EIS message ends with an EOM flag so that regular communications can resume rapidly.

VI. THE EIS HEADER. The largest single part of the EIS protocol, is also one of the shortest parts of the EIS message. The EIS message always begins with a data message sent at 520 bits-per-second. This message is in standard ASCII. The nonstandard baud rate results in added security and non standard mark and space frequencies of 1562 Hz and 2083 Hz. The header is repeated three times, separated by 1/2 second pauses, to assure reliable reception. If it is to be retransmitted, it would also be regenerated. Individual decoders are programmed to perform the automatic, or manual rebroadcast functions. The EIS FSK data is transmitted at a minimum of 50% modulation of full channel modulation restraints, with the exception of those channels maintained at some other level as provided for and specified by the state EIS plan. Any other modulation occurring in the EIS passband must be suppressed by at least 20db below the EIS FSK data during the transmission of the EIS message. The EIS Header consists of six parts:

1. The PREAMBLE. The EIS preamble consists of a string of 10101011...(8 bit bytes) for a period of 1/2 second (500ms or 32 sync characters). This allows decoders to detect the presence of the EIS message, and prepares automatic gain controls and other circuits for the message. The decoders will be able to scan all of their channels inside of the 500ms preamble time, and lock onto the channel with EIS signaling. As a result of the scanning process, decoders are virtually assured of receiving the EIS message arriving by the shortest and presumably best path (secondary, or repeated sources will retransmit the EIS message with some delay, and thus the most primary source, having already been selected, will have the attention of the decoder).

2. The ENCODER IDENTIFICATION CODE. Following the preamble, a four alpha-character serial number code is transmitted, then a "-" character, and a three alpha-character identification of the activating organization, followed by a "-" character.

The Serial number identification "ZCZC" is reserved for all existing and future weather radio WR-SAME encoders. All other encoders will have an individual and unique identification. Manufacturers will be assigned a range of identification characters. For example, a given manufacturer may be assigned TAAA through TZZZ which represents 17,576 unique identification codes.

The three character activating organization codes are assigned in such a manner that each agency has a unique code. Initially, National level organizations would be assigned the range USA through USZ, and states would be assigned a group beginning with the two letter US Postal state code, and followed by the range A-Z. For example, the Colorado State EOC might be designated the identification "COE" as part of the state plan. The large number of other available codes would be assigned as needed. The characters "WXR" are reserved for NOAA weather radio systems utilizing the WRSAME system.

3. OPTIONAL TEXT VERSION OF ENCODER IDENTIFICATION. This block is reserved for a simple sentence, such as "COLORADO STATE EOC, DENVER." By using plain text, it would permit rapid display of the source of the activation, without the requirement of "looking up" the agency from a data base of encoder serial numbers. Using this code would require a modification of the WR-SAME protocol to allow the additional block of information, or some WR-SAME receiver designs would have difficulty decoding this block.

4. The EVENT (TYPE OF ACTIVATION) CODE. The next three alpha characters indicate the type of emergency message. These are preceded and followed by a "-" symbol. The table is as follows:

National Codes

NLA National priority activation
NLS National EIS statement
NLT National test

Weather Service Codes

TOA Tornado watch
TOR Tornado warning
SVA Severe thunderstorm watch
SVR Severe weather statement
SVS Severe weather statement
SPS Special weather statement
FFA Flash flood watch
FFW Flash flood warning
FFS Flash flood statement
FLA Flood watch
FLW Flood warning
FLS Flood statement
WSA Winter storm watch
WSW Winter storm warning
BZW Blizzard warning
HWA High wind watch
HWW High wind warning
HUA Hurricane watch
HUW Hurricane warning
HLS Hurricane statement/update
LFP Service area forecast
BRT Combined/special routine broadcast

Local/State Codes

CEM Civil emergency warning
CES Civil emergency statement
TRA Traffic announcement
TRE Traffic emergency
TRT Traffic authority test
EKS Earth quake statement
EIS EIS administrative information
ENA Environmental statement
ENW Environmental warning
EVC Evacuation statement/information
FRS Fire statement
FRW Fire warning
FRT Fire/forest service test
STA State priority activation
STS State EIS statement
STT State test
OPA Operational area priority activation
OPS Operational area EIS statement
OPT Operational area test
PLA Police priority activation
PLS Police EIS statement
PLT Police test
SHA Sheriff priority activation
SHS Sheriff EIS statement
SHT Sheriff test
MLA Military priority activation
MLS Military EIS statement
MLT Military test
XXA Local assigned additional entry point priority activation
XXS Local assigned additional entry point EIS statement
XXT Local assigned additional entry point test
YYA Local assigned additional entry point priority activation
YYS Local assigned additional entry point EIS statement
YYT Local assigned additional entry point test
ZZA Local assigned additional entry point priority activation

ZZS Local assigned additional entry point EIS statement
ZZT Local assigned additional entry point test

General Codes Available To All
DMO Practice/Demo
CCC Closed circuit news/announcer information
ADR Administrative message

Use of state and local codes are defined by the state EIS plan.

The Event Code is used by the decoder to determine what action is to be taken, and how the EIS message is to be routed. Individual decoders can be programmed (using a simple look-up table) to take the three letter Event Code and display or print a plain text message.

5. The COUNTY CODES. Each EIS message contains the six digit "FIPS" code for the affected counties. Should an area smaller than a county be desired, the first digit, which would always be a "0" in the FIPS code, would be replaced by a number 1-9 which would represent a portion of a county as defined by the state plan. Each county code is separated by a "-" symbol.

The county code is used by the decoder to determine if the EIS activation is for an area that is served by the media outlet or communications channel. It is part of the information used to determine the routing of EIS messages.

For example: 139173-039051- could indicated Andersonville, Wood County, Ohio; and Fulton County, Ohio as the area the EIS message is targeted to.

In addition to the FIPS codes, there are wide area codes as follows:

099999 All of the United States of America
0xx999 All of a given State code, where the first three digits represent the state FIPS code.

6. The DELTA TIME statement. This indicates the "life expectancy" length of the emergency or warning. This information also signals the end of the Header, and the beginning of voice or data information. The following are available codes.

+00- No time specified
+30- One half-hour
+45- 45 minutes
+60- One hour
+61- More than an hour

The delta time information can be used by automated devices to determine when to repeat EIS messages.

Following the header, voice or data may follow. It is assumed that the message is voice unless or until the character string "-DDDD- DDDD-DDDD-" is sent, indicating the presence of data.

Data is transmitted using the same FSK format as the EIS header.

The time limit for a voice message is 90 seconds, at which time the decoders would "time-out," and reset.

There is no predetermined time limit for data communications, however the lack of data would reset the EIS encoder.

The character set "-NNNN-NNNN-NNNN-" is used to indicate the end of the EIS message. Following the end of message signal, the communications channels are returned to normal operation.

PRACTICAL EIS OPERATION:

The EIS is composed of hardware and software. The hardware is the signaling scheme and the basic architecture of the system, the software is the State EIS Plan. Similar to the State EBS Plans, The plan assures that each entry point has the proper codes available for transmission, and enough communications paths to reach each decoder served.

The Plan would cover both how the EIS message was sent from the activating authority, and how it was relayed to the electronic media. Further, the Plan would determine which media outlets monitored which communications channels.

The Plan would restrict encoders, so that only relevant areas and Event Codes could be transmitted. The plan would also determine testing schedules.

As a practical matter, the NOAA weather radio stations could serve a larger role than they now do, in that as part of the State EIS Plan, they could serve not only as an origination point, but as a distribution point, or relay for EIS messages generated elsewhere. A typical NOAA weather radio station's decoder might monitor the PEP station, wire lines from local authorities, an RPU frequency set aside for this purpose, and public safety radio channels. For redundancy, the NOAA weather station might also retransmit an EIS message originating with the NWS via the NOAA weather radio, and an RPU channel or public safety channel.

With broadcasters, there is the opportunity to use the over-the-air signal as major distribution points. An automated, wide coverage area FM station that agrees to carry all priority and EIS statements can become a major part of the EIS distribution plan.

Other opportunities also exist. For example, "super stations" carried via satellite can be used as part of a state's distribution plan. Likewise, mountain-top communications repeaters can be used to distribute EIS messages with the agreement of the operators.

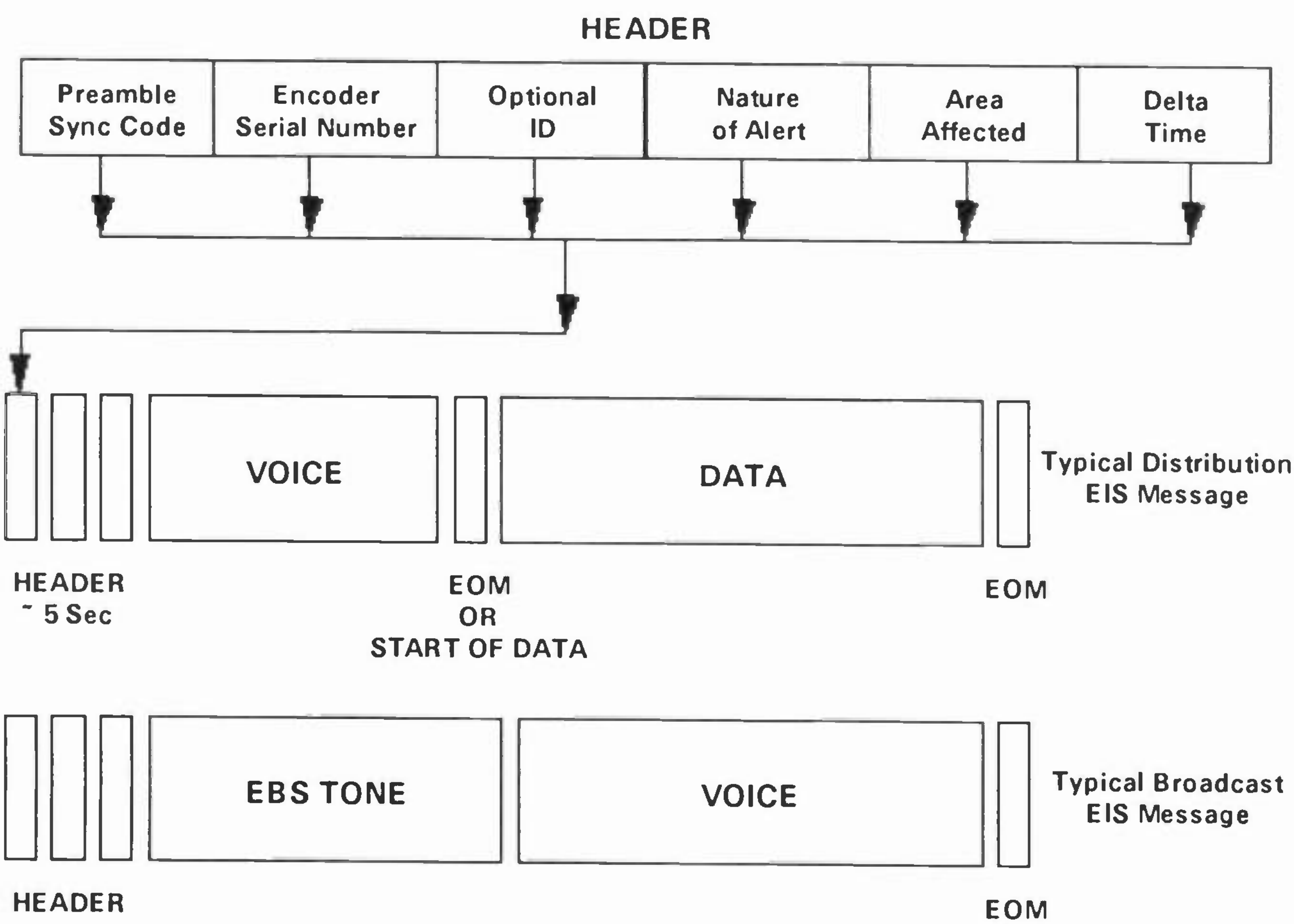

Figure 1: Time Line Construction of EIS Message

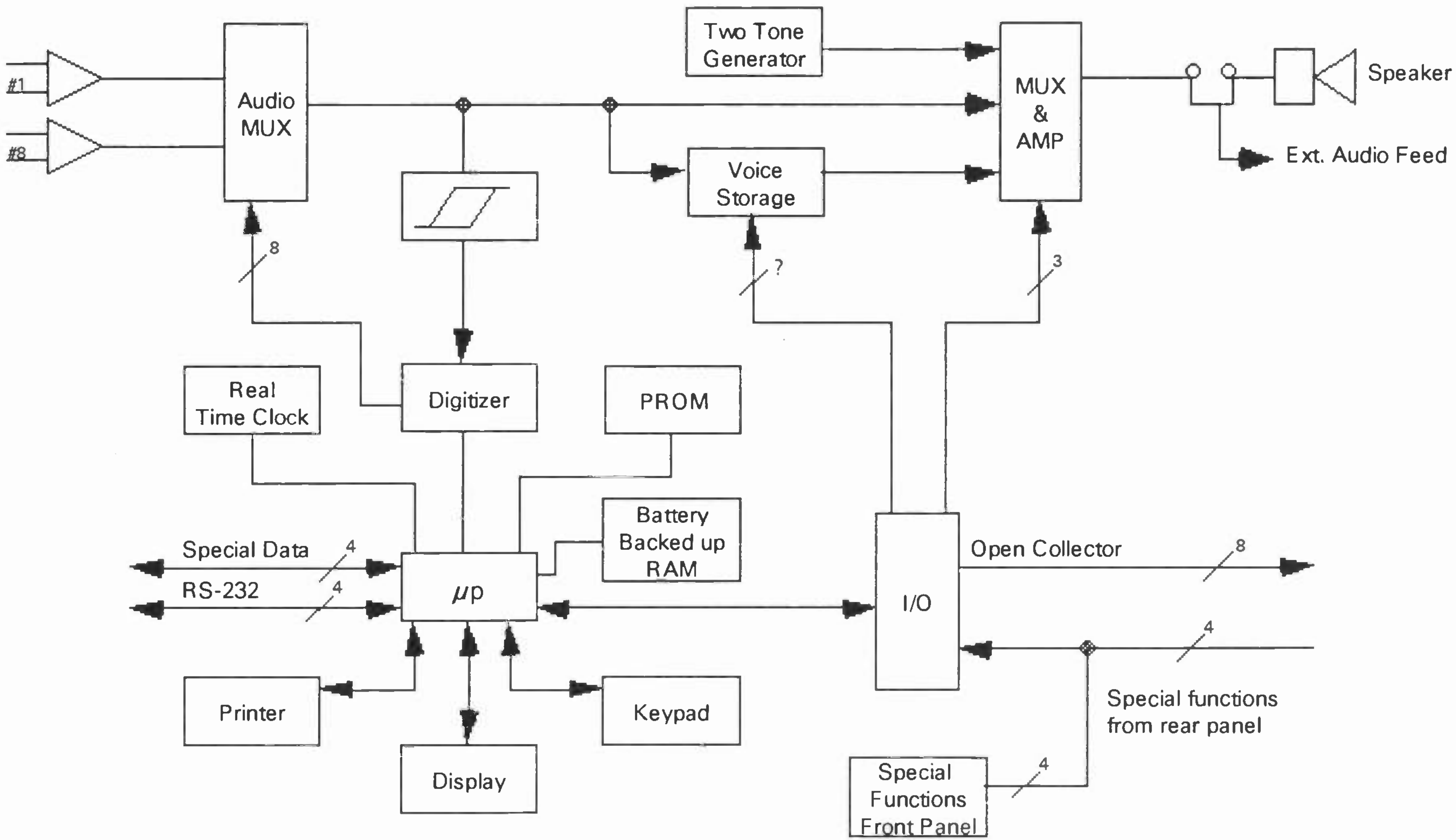

Figure 2: Block diagram of TFT's version of an EIS node "Manager", approach to the EIS hardware

SBE DAY
CONTRACT ENGINEERING WORKSHOP*

Tuesday, April 20, 1993

Moderator:
Chip Morgan, Chip Morgan Broadcast Engineering, Folsom, California

Panelists:
Barry Victor
The Victor Group
Los Angeles, California
Chris Imlay
Booth, Freret and Imlay
Washington, District of Columbia

*Papers for this session were not available at the time of publication.

DATA BROADCASTING: RADIO

Wednesday, April 21, 1993

A PRELIMINARY ANALYSIS OF THE PROSPECTS FOR UTILIZATION OF SECONDARY CAPACITIES IN THE COMMERCIAL FM RADIO AND TV CHANNELS FOR IVHS DATA COMMUNICATION NEEDS
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COUPONRADIO: PROFIT POTENTIAL OF RBDS
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RDS/RBDS INSTALLATION GUIDELINES AND FUTURE INTERFACE APPLICATIONS
John D. Casey and Robert McCutcheon
RE America, Inc.
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***PRACTICAL FIELD EXPERIENCE WITH RBDS**
Mark Krieger
WGAR-FM
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NHK'S HIGH CAPACITY FM SUBCARRIER SYSTEM
Osamu Yamada, Masayuki Takada, Toru Kuroda and Tadashi Isobe
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*Paper not available at the time of publication.

A PRELIMINARY ANALYSIS OF THE PROSPECTS FOR UTILIZATION OF SECONDARY CAPACITIES IN THE COMMERCIAL FM RADIO AND TV CHANNELS FOR IVHS DATA COMMUNICATION NEEDS

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Abstract: In this paper, in addition to briefly describing the IVHS application and its communication needs, the authors summarize their investigation of the commercial FM Radio and TV Broadcast channel structures, available capacities, possible data rates, and the approaches by which IVHS and other services can take advantage of the available broadcast conduits. This also makes good business sense to both parties - the IVHS community and the broadcast service vendors.

1.0 Introduction

Commercial broadcasting systems have recently been looked at with much interest and optimism for the data transmission capacity available in their channel structures. Such sharing is not only more spectrum efficient but also offers reduced development and deployment time, and lower costs for IVHS, as a mechanism for an evolutionary architecture. Carrying traffic data on subcarriers will produce revenue for broadcasters.

In Europe, the Radio Data System (RDS), which uses the local FM radio broadcast infrastructure, is already standardized and is being widely deployed. Also, the U.S. is in the process of adapting similar standard called the Radio Broadcast Data System (RBDS). In the US, efforts are underway to adapt the FM subcarrier technology within Subsidiary Communications Authorization (SCA) for transmission of IVHS data. In the commercial Television world, the Vertical Blanking Interval (VBI), and aural subcarriers, such as the Second Audio Program (SAP), present interesting prospects for utilization by IVHS and other newly emerging wireless services.

1.1 Intelligent Vehicle Highway Systems (IVHS)

The Intelligent Vehicle Highway Systems (IVHS) program is an important part of the Intermodal Surface Transportation Efficiency Act (ISTEA) of 1991. The ISTEA makes provisions for a substantial amount of funding to support development and implementation of IVHS on the nation's road network. The aim of the IVHS program is to apply advanced concepts and technology in the areas of communications, control, navigation and information systems to reduce highway congestion, improve highway safety, render highway traffic more compatible with the environment, and improve U.S. international competitiveness.

Communications, especially wireless communication, is a crucial aspect of IVHS implementation. There are six major system areas within IVHS: Advanced Traffic Management Systems (ATMS); Advanced Traveler Information Systems (ATIS); Commercial Vehicle Operations (CVO); Advanced Public Transportation Systems (APTS); Advanced Rural Transportation Systems (ARTS); and, Advanced Vehicle Control Systems (AVCS). A brief description of these system or technology areas follows:

Advanced Traffic Management Systems (ATMS): permit real-time adjustment of traffic control systems and variable signing for driver advice. Their application in selected traffic corridors has reduced delay, roadway link travel times, and accidents. ATMS is being implemented using coordinated traffic signal systems, video surveillance of corridors, ramp metering, automated toll collection, and variable message signs (VMS).

Advanced Traveler Information Systems (ATIS): deal with the acquisition, analysis, communication, presentation, and use of information to assist the surface transportation traveler in moving from origin to destination in the way which best satisfies the traveler's needs for safety, efficiency, and comfort. Travel may involve a single mode or linked, multiple modes. ATIS let travelers know their location and how to find services.

ATIS permit communication between travelers and ATMS for continuous advice regarding traffic conditions and alternate routes. Additionally, ATIS provides the driver with warnings regarding road safety.

Commercial Vehicle Operations (CVO): expedite deliveries, improve operational efficiency, improve incident response, and increase safety. CVO makes use of ATIS features critical to commercial and emergency vehicles. A primary goal of CVO is to reduce regulatory burden and inefficiency. Many of the technologies related to CVO are already available in the marketplace. Automatic Vehicle Identification (AVI) devices are used in several locations to allow the electronic transfer of funds so travelers can pay tolls without stopping. Global Positioning System (GPS) and Loran-C technologies are available to track the location of individual vehicles for fleet management. Weigh-in-Motion (WIM), combined with Automatic Vehicle Classification (AVC), is available to sort vehicles for weight inspections. On-board computers are available to monitor truck performance.

Advanced Public Transportation Systems (APTS): work in conjunction with ATMS and ATIS to provide mass transportation users and operators (e.g., buses, vanpools, high-occupancy vehicle (HOV) lanes, carpools, taxi cabs) with up-to-date information on status, schedules, and availability of public transit systems. Automatic vehicle location and monitoring systems will provide information to improve fleet management and better inform riders of their connections. Electronic fare media will reduce the inconvenience of cash handling, provide new marketing data, and integrate third party billing for transit services. New HOV priority schemes using IVHS technologies will be devised and monitored automatically to enforce HOV facility use.

Advanced Rural Transportation Systems (ARTS): ensure safety on the nation's rural roadways, facilitate highway maintenance, and provide navigational aids to tourists. ARTS is a relatively new concept covering IVHS applications in the rural areas.

Advanced Vehicle Control Systems (AVCS): are vehicle- and/or roadway-based electro-mechanical and communications devices that enhance the control of vehicles by facilitating and augmenting driver performance and ultimately, relieving the driver of most tasks on designated, instrumented roadways.

2.0 IVHS Communications Requirements

Overall, different communications technologies are found suitable for these various IVHS areas. Within the envelope of wireless communications, which is the most important class of communication mechanisms for IVHS, a wide range of radio frequency bands from 50 MHz to a few tens of Gigahertz are suitable for applications in the six system areas. In this section, we have outlined the *high-level* communication needs of IVHS.[1] It represents the general spectrum requirements that are applicable to the different areas within IVHS.

2.1 Infrastructure-to-Vehicle

The moving automobile may employ a number of different techniques to satisfy its various IVHS communication needs. It is easily seen that the vehicle-to-infrastructure (and vice-versa) communication link has to be tetherless and requires suitable spectrum. Mobile radio systems work most efficiently in the 50-1000 MHz range. Information to vehicles may be most efficiently conveyed by broadcasting. For example, Highway Advisory Radio (HAR), Advanced HAR, Radio Data System - Traffic Message Channel (RDS-TMC) are already deployed in the field. Again, depending on the particular implementation, roadside low-power, local area transmitters along with wide-range high-power transmitters could be used. Each approach, nonetheless, requires radio frequency arrangements and coordination/integration with/into the full IVHS.

The Subcarrier Communications Authorization (SCA) subcarrier channel, very much like RDS, is a suitable option for infrastructure-to-vehicle communication, in that it makes use of the 'spare' capacity available in the baseband commercial FM Broadcast systems. Initial IVHS implementation will most certainly adopt such an approach to provide first user benefits, quickly and inexpensively.

2.2 Vehicle-to-Infrastructure

The nature of vehicle-to-TMS/TIC (Traffic Management System/Traffic Information Center) communication requires only a low data rate. This inbound mobile communication is needed for inquiries which are relatively short (message) data-strings, and for reporting incidents, link times and destinations to the TIC. Database queries, for example yellow-pages, may be tied into privatized services; nonetheless, only the downstream data flow, in response to such inquiries may be heavy and will dictate high data rate communication links.

It should be noted, however, that some proposed IVHS architectural concepts require the transmission of information on intended routes, which would impose a much higher data rate requirement.

Vehicle-to-infrastructure communication requirements may expand to require a more sophisticated and two-way IVHS implementation providing a richer set of services. Wide-area-radio, infrared/microwave beacons, and perhaps, a digital cellular infrastructure may be the appropriate candidates for evaluation in order to meet this category of IVHS system objectives.

AVI application needs are for short range communications, and can use frequencies at or above 1 GHz. These systems can use either mode of communication - one-way or two-way.

2.3 Inter-Vehicle

Vehicle-to-vehicle communication requirements for Advanced Vehicle Control Systems (AVCS) dictate high data rates and large bandwidths. This application can be operated in the range well above 1 GHz. However, a number of questions need be answered before one can attempt to design such a system. For example, what bands must be used for different applications to avoid interference among platooning, collision avoidance, and overtaking/blind-spot-alert systems. The questions regarding range and power-levels, bandwidth allotment (as a function of geographical location or zones), and communication protocols are but a few that need to be thoroughly examined.

2.4 Point-to-Point between Infrastructure Nodes

Communication links among TMSs/TICs may be implemented via dedicated fiber optic (FO), microwave or T1 links. Remote monitoring (for example, by video cameras) may require microwave or satellite links in such situations when FO/coaxial cable is not a viable alternative. Mobile TMSs, if ever designed and deployed, will always require radio links.

2.5 Infrastructure-to-Pedestrian

For APTS applications it may be necessary to broadcast relevant traffic information - for example, arrivals of public transit vehicles and parking availability etc. - to the motorists as well as to the "pedestrians". Such a communication requirement can, perhaps, optimally be met by a "local" broadcasting system or made available via a public facility, such as a Personal Communications Network (PCN).

3.0 Prospects for Utilization: Existing Broadcast Infrastructure

Commercial broadcasting systems have recently been looked at with much interest and optimism for the data transmission capacity available in their channel structures. Such sharing is not only more spectrum efficient but also offers reduced development and deployment time, and lower costs for implementing IVHS, as a mechanism for an evolutionary architecture.[2] Carrying traffic data on subcarriers will also produce revenue for broadcasters.

3.1 FM Radio Broadcasting

In Europe, the Radio Data System (RDS), which uses the local FM radio broadcast infrastructure, is already standardized and is being widely deployed. However, the capacity limitations of RDS restrict its use in IVH S that are being designed, developed and field-tested. The overall RDS channel capacity is 1187.5 bps, but only about 300 bps is available for "new" uses.

In the United States, efforts are underway to adapt the FM subcarrier technology in Subsidiary Communications Authorization (SCA) channel for transmission of IVHS data. The results of a recent investigative study carried out on behalf of the Federal Highway Administration (FHWA) suggests basic departures are needed from the RDS design to better suit the U.S. situation and provide increased benefits.[3] Furthermore, efforts by the Federal Highway Administration (FHWA) are underway to develop and test a prototype based on a conceptual design which should provide a much greater data rate subcarrier channel on commercial FM Radio broadcast systems. The new system, called SCA Traffic Information Channel (STIC), has a much greater information capacity than other approaches, enabling it to support more IVHS services. The STIC system is also flexible and scalable so as to be able to adapt to more sophisticated phases of IVHS implementation.

3.1.2 SCA Traffic Information Channel (STIC) Concept for the United States

By the mid-1960s, the FM broadcast stations in the U.S. used subcarriers to transmit subsidiary audio programs that provided background music in restaurants and shops. This type of transmission is governed by the Subsidiary Communications Authorization (SCA) under CFR 47, Part 73 by the Federal Communications Commission (FCC). Such SCA systems, however, were not found suitable for use by European broadcasters because there was unacceptable cross-talk from the SCA program into the main program due to the fact that FM channels in Europe are only 100 kHz as compared to 200 kHz in the U.S. Critics have noted the "waste" of FM broadcast spectrum in the U.S. because it is 200 kHz wide, while Europe "is getting by" with only 100 kHz. However, this circumstance could turn out to be in favor of the U.S. But the wider bandwidth will aid the IVHS by facilitating design of a superior STIC system for the U.S. which offers greater data rate capability than, but compatible with, RBDS.

3.1.2.1 STIC System Features

This particular design, although employing relatively straightforward coding and modulation schemes, provides a total system data rate of about 18.8 kbps. In order to maintain compatibility with RBDS, and, perhaps, with

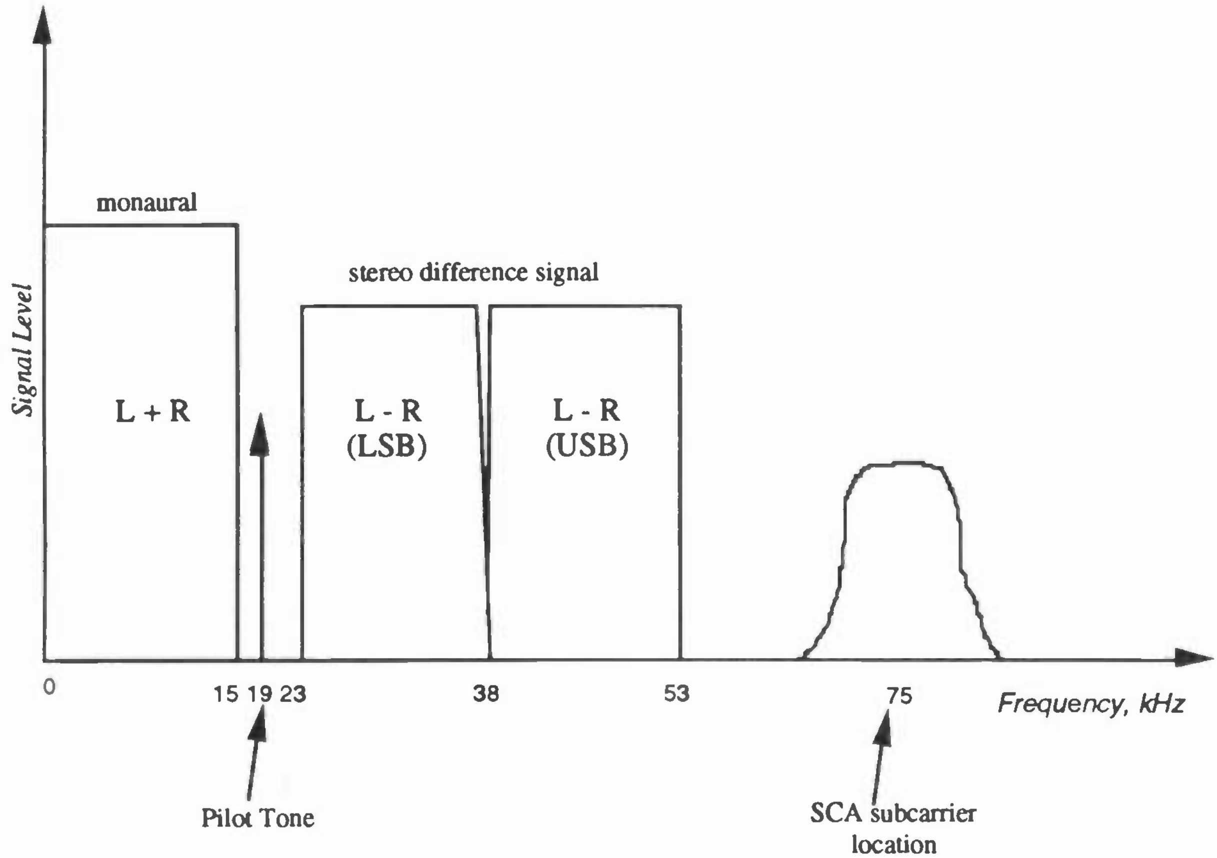

Figure 1: FM Baseband Format Under the STIC System

subcarriers at 92 kHz, STIC will use a 75 kHz subcarrier frequency. (See Figure 1.) This subcarrier at 75 kHz will be modulated by the digital traffic data-stream, and then used to frequency modulate the main carrier to a level corresponding to a modulation index of up to 20 percent.

Communication Capacity and Candidate IVHS Applications

It is estimated that even with full ATIS/ATMS implementation, a downlink data rate of about 18.8 kbps would be enough to handle required services, including updating the in-vehicle link-time database with acceptable periodicity.

LINKT message group, containing realtime data on roadway link travel times, will provide the updates for the link travel times to the vehicles as and when necessary. They will also be used for periodic refreshing of the basic link-time database resident in the vehicles (e.g., with a periodicity of once every five minutes).

EMERG message group, containing emergency messages for the motorists, will broadcast emergency-related data when the situation demands.

TMC data group will contain standard RDS-TMC messages.

INCLOC data group, containing intersection-level location information associated with a traffic incident, will be transmitted whenever a TMC group transmits incident-related information that has associated location information to be broadcast.

TRNST message group, containing transit schedules and related parking availability information, will be used to convey desirable schedules of, and connections to, the public transit system, both surface and subsurface systems.

DGPS data group, containing timing and location related correctional data associated with the use of the Global

Positioning System (GPS), will be utilized to broadcast the correctional (time and location) parameters computed by the differential GPS station for the area.

Service Range

One of the design goals of the STIC system was to maintain the equivalent coverage area as that for the main stereo entertainment programs. Although the power imparted to the STIC signals is lower than the main program, the digital nature of the STIC information makes the Signal-to-Noise Ratio (SNR) requirements less stringent than the analog audio program. Calculations reveal that with the help of error correction options available to any digital scheme, the Bit Error Rate (BER) performance can be improved to achieve the same service range for STIC users as that for the main program users, while, at the same time, providing net *user* information data rate of about 8 kbps.

3.2 Television Broadcast Systems

In the commercial television world, the Vertical Blanking Interval (VBI), and aural subcarriers, such as the Second Audio Program (SAP) channel, present interesting prospects for utilization by IVHS and other newly emerging wireless services. Broadcast TV channels offer two basically different outlets that may support data communication for IVHS.

3.2.1 Vertical Blanking Interval (VBI) Data Systems

In the United States, the standard television signal broadcast comprises of 525 horizontal lines for a picture frame. The picture frame is divided into two fields; each field has 262.5 lines. The lines of the two fields are interlaced on the picture frame --- one at a time alternating from each field to make up the line-sequence of the picture. The first 21 lines of either field makes up what is called the Vertical Blanking Interval (VBI), the black strip seen on the screen when the picture is made to roll vertically. The VBI is part of the composite video signal (see Figure 2) but has no information in it.[4]

The TV receiver requires the first nine lines for timing setup. However, the lines 10 to 18 are not needed by the current-day TV receivers and may be used for transmission of various kinds of data, which in turn could be used to provide a wide variety of services. Presently, line 21 is frequently used to convey the closed captioning signals, and line 19 to transmit the Vertical Interval Reference signal. This method of data transmission by using the VBI was in the past known as teletext or videotext, but now the standard way of referencing it is VBI data broadcasting.

Recent legislation though has mandated the use of line 21 for conveying closed captioning information. Furthermore, it seems almost certain that the line 19 will be reserved (or allocated) to transmit ghost cancellation signals --- standardization efforts are almost completed.[5] Finally, line 20, although not mandated, is widely used for program identification and program rating purposes, and therefore, may not be available for IVHS. Thus, remaining lines 10 to 18 in each field --- 9 lines in total per field --- are available for IVHS and other applications.

Capacity: Each horizontal line of a TV picture frame is a 'timeslot' lasting approximately 63.5 microseconds. According to the North American Basic Teletext Specifications (NABTS), each of these lines can carry up to 288 bits; out of which 64 bits are essential overhead and addressing bits. This leaves us with about 224 bits for each of the 9 lines available in every field. Now, there are 30 picture frames or 60 fields every second yielding a 'gross' data rate of about 120 kbps. With NABTS specified forward error correction algorithm, the available information data rate would be in the vicinity of 90 kbps. Of course this assumes stacking or chaining the lines to combine their individual capacities, and the technology for achieving this exists.

For mobile applications, as is the case with IVHS, we estimate that about 50% of the data transmission will be consumed by the sophisticated multi-layered coding schemes to counter the destructive effects of multipath and fading phenomena. In spite of that the data capacity remaining for conveying pure information will be 40-50 kbps on a single TV channel. Nevertheless, because of the mostly mobile nature of IVHS applications, it appears that the VBI capacity may be more suitable for 'stationary' IVHS functions; for example, periodically distributing realtime updates on traffic situations controlled by various Traffic Management Centers (TMC) in different cities and regions to make available every TMC with nationwide 'realtime' data for route guidance and trip planning.

The above total capacity may be shared among various services that require one-way broadcasting of data. For example, the different IVHS applications or services as listed in earlier sections on FM SCA Traffic Information Channel system (Sec. 3.1.2.1) could subdivide, perhaps dynamically, the share of road traffic related services. There are some equipment suppliers with products in the market which are somewhat expensive at this point in time, hopefully, with market expansion and refinement of design their prices will come down.

3.2.2 Aural Subcarriers

A TV transmission station is really two "associated" broadcast stations with separate transmitters for the visual and aural signals. The TV aural transmitter is actually just an FM station with somewhat different technical characteristics. As such, subcarriers can be supported as

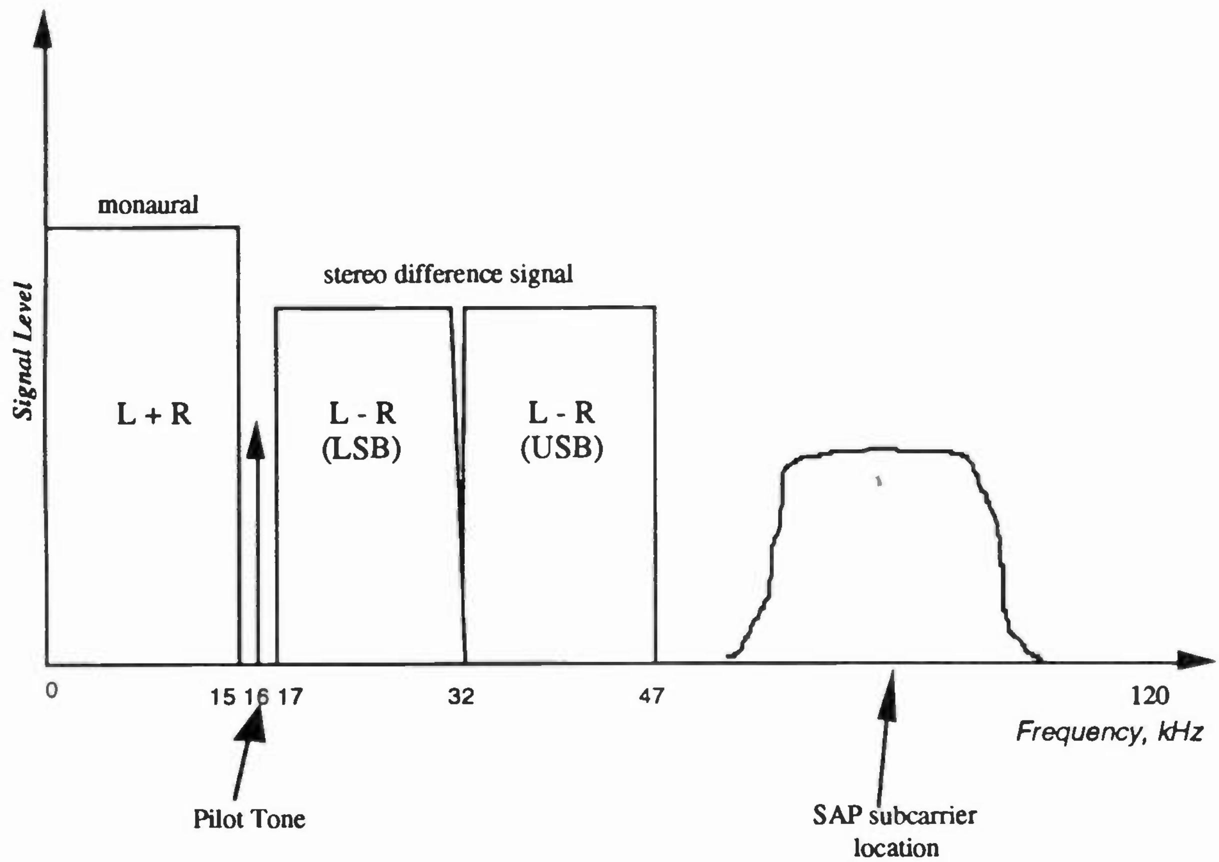

Figure 2: Typical TV Aural Baseband Showing Second Audio Program Channel

with conventional FM. The range in most cases is as good as in case of conventional FM radio transmission.

The professional (PRO) channel is a fairly narrowband channel used by the TV station to deliver special instructions to the news reporters at remote locations. Consequently PRO channels are not available or have little value to service providers requiring much higher bandwidth (or data rate) --- such as, the IVHS community.

The other standard TV aural subcarrier is called the "Second Audio Program" (SAP) channel. It is infrequently used for some of the same services as FM SCA, and occasionally to broadcast a running commentary of the main TV program for the visually handicapped. The SAP channel has a much wider bandwidth than SCA subcarriers, although the basic technology is identical. SAP receivers are only available as a separate device (i.e., not built into a TV set) from one manufacturer at present. Since the bandwidth available on a TV aural signal is large, more than one wideband subcarrier can be accommodated. Even in those cases where a TV station is using SAP, another subcarrier can be added for traffic data.

On a monaural TV channel, as much as 104 kHz, from 17 to 120 kHz of the aural baseband, may be available for one or more SAP channels. In the case of a stereo TV aural baseband, 74 kHz, from 47 to 120 kHz, could be available for IVHS and other applications. Figure 2 depicts a typical SAP channel location in a stereo aural TV baseband.

As compared to the FM SCA subcarriers for mobile applications, the TV aural channels provide more bandwidth which results in a relatively higher data rate. However, the multipath and fading environment remain to be taken care of in any mobile system design. However, one has greater overheads available to employ more sophisticated coding schemes to mitigate the undesired effects of multipath and fading.

A commercially available service utilizes SAP channel to transmit road traffic information for users --- mobile or stationary --- to receive the audio information with the help of a unit that costs about $129.

4. Recommendations for Future Work

In view of the above discussions on the feasibility of using the TV broadcasting industry's VBI and the aural SAP channels, it is desirable that prototype projects be planned to investigate in greater detail the design and commercial aspects of the two prospective technology candidates to bear the sizeable IVHS communication requirements.

The authors also believe it is critical for IVHS communication requirements to be considered in the standards process regarding new broadcast technologies maturing for commercial deployment --- namely, Digital Audio Broadcasting (DAB) and High Definition TV (HDTV).

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COUPONRADIO: PROFIT POTENTIAL OF RBDS

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Radio broadcasters have long been anxious, yet remarkably patient, in their hopes that a new technology or a good idea would come along that would dramatically increase their profits, while giving the broadcast industry new excitement and lift.

This paper will outline and explain how CouponRadio (a patented technology coupled to a comprehensive business plan), can accomplish this exact goal. CouponRadio utilizes the United States Radio Broadcast Data System (RBDS); transmitting data on a broadcasters FM 57khz subcarrier; and specially designed radios which will not only receive and display changing messages, or portions thereof, but will, more importantly, allow the listener, utilizing one of the radios special CouponRadio features, to selectively save the messages of their choice, in its entirety, onto removeable creditcard like memory devices which can be left in the radio for viewing later, or removed from the radio to generate valuable coupons at CouponRadio kiosks.

Introduction

The United States economy, once fueled by manufacturing oriented industries, is now recognized as an economy dependent upon technology, information and services. CouponRadio Inc.'s patented process and plan is offered as a way the broadcast industry can greatly improve its abilities as a group to effectively compete with other media in this new economy.

The CouponRadio goal is to use **technology** to provide the radio listener with more **information** and better **service**, by making what is heard over the radio, unforgettable.

Long ago, radio literally owned the attention of all available listeners. Radio had no competition. Today, so much has changed, yet, nothing has changed. So much has changed, in that the population has exploded along with technology. Today, electronic media of all kinds fiercely compete for a share of the audience and advertisers, once monopolized by radio.

Today, to the dismay of listeners, advertisers, and broadcasters, nothing has changed. Nothing, in that, with all the great technological achievements made over time, radio still remains the same medium it was generations ago. An audio-only medium, whereby those who listen would have difficulty remembering important information they heard broadcast over the radio.

More than just adding another feature to a radio, CouponRadio recreates an entire broadcasting industry, making it interactive, and far more profitable through the use of radio data technology, memory and coupon machines.

CouponRadio

CouponRadio operates under the premise that all radio text messages **(INFOMESSAGES)** are valuable and should be saved. Limited to 64 characters of text, the coupon formatted infomessages, are stored in a computer data base by the broadcaster, at the radio station. Each infomessage is given a specific access code, associating it to a particular program piece or advertiser. As the regularly scheduled programming is being aired, its corresponding coupon infomessage is being simultaneously accessed and sent from the computer's RS232 port to the data encoder.

Example: On 4/10/93 @ 11:43am, the song "Unforgettable," by Natalie Cole, was broadcast simultaneously with this infomessage: **Unforgettable Natalie Cole Elektra-61049.** On 4/10/93 @ 2:15pm, the advertisement for Manhattan Motors was broadcast with this infomessage: **Manhattan Motors 500 W. 59 St. 212-582-7700 $100 Off Best Price.**

The encoder instantly transmits the infomessage over the stations 57khz subcarrier, to a CouponRadio equipped receiver. The new infomessage will appear on the radio receiver's alphanumeric display, replacing the preceding message.

Shown in **Fig. 1,** the CouponRadio prototype, includes a 16 character alphanumeric display, provision for a RadioCard, a memory button, a scroll button, and 3 indicator lamps for: memory full, insert RadioCard, and broadcast/recall mode. A typical European auto receiver is used to provide the RBDS signal. These CouponRadio features can easily be made a part of any radio receiver.

At the push of the MEMORY button, the radio receiver will allow the listener to instantly save the valuable coupon formatted infomessage currently being displayed. Each time a listener presses the memory button, the entire infomessage is saved, even though only a small portion of the infomessage is visible, due to the size restraints of the alphanumeric display confronting receiver manufacturers.

Figure 1: CouponRadio Receiver Prototype

We have found that the use of a 16 character horizontal display in our auto receiver prototype has been small yet very versatile. It has no affect of limiting ones vocabulary by forcing the use of small words, and also enables any infomessage to be read back quickly and easily without the need to cut off parts of words. Our testing of other size displays showed that the 12 character display was also effective in reading out infomessages. Both the 12 and 16 character displays offer the broadcaster more room to show additional station information. Eight character horizontal displays do not work well when trying to read back infomessages, and are not recommended in any configuration.

Another feature of CouponRadio is the radio broadcast industry's historic use of the RadioCard, a creditcard-like memory device, with a read/write memory chip built into the plastic. All valuable infomessages, saved by the listener, get stored on this unique card. Each time a listener desires to save an infomessage, pertaining to what is being heard on the broadcast, the pressing of the memory button will cause the information to be immediately stored on the RadioCard. When the memory button is pushed, the CouponRadio equipped receiver will further decode the RBDS data signal, and store onto the RadioCard, the time of day, date and station call letter information. This valuable information will be printed on CouponRadio coupons.

CouponRadio also proposes that each time the receiver's memory button is pressed, the word AUTO, WALK or HOME, describing receiver type, is additionally stored on the RadioCard, and later printed, for coupon redemption purposes.

At a safer or more convenient time, the listener can switch the CouponRadio receiver into the RECALL MODE by pressing a SCROLL button. The alphanumeric display will then read out the first segment of one

UNFORGETTABLE

message. The first segment is equal to the number of character positions physically available on the display. In this mode, the alphanumeric display will only read out the coupon formatted infomessages stored on the RadioCard.

Each time the scroll button is pressed, the alphanumeric display will read out the first segment of the next message in sequence. In

MANHATTAN MOTORS

the recall mode, once the first segment of a desired message appears on the display, the memory button can be used to read through the remainder of the message.

500 W 59TH ST

212-582-7700

$100 Off Best Price

This feature enables the listener to scroll through messages quickly, until the desired infomessage is reached, rather then reading through every message. To advance to a different infomessage, the scroll button is pressed again, until the desired infomessage is reached.

The memory ability and scroll features of CouponRadio make the receivers extremely safe in automobiles. At a push of a button, the

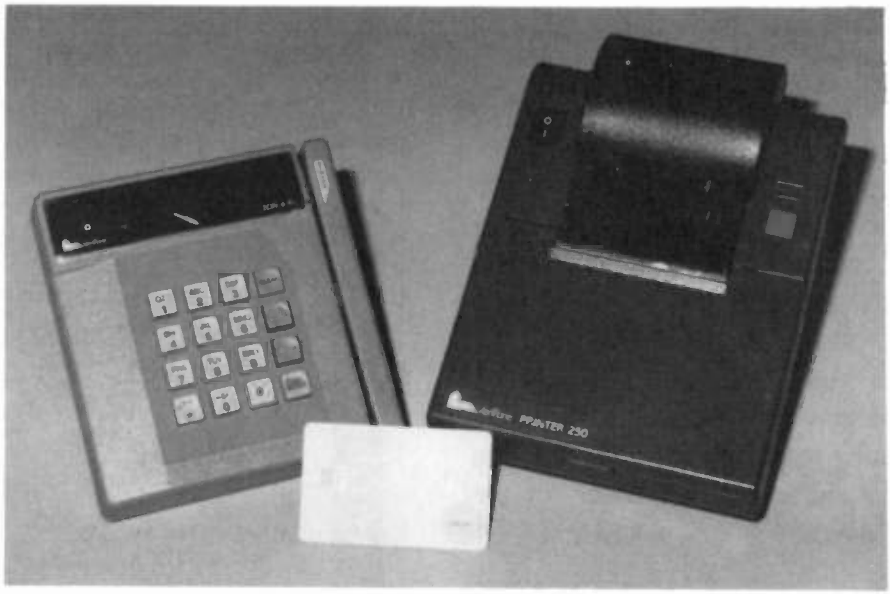


Figure 2: Typical Credit Card Verification Terminal, Printer and RadioCard.

infomessage can be saved so as not to encourage viewing.

Coupons

The CouponRadio System provides for two types of coupon printers, for different applications.

The first CouponRadio printing system makes use of a small counter-top type, dot matrix printer, a familiar machine, used regularly by millions of merchants for credit card authorization and receipt issuance. **Fig. 2.** The CouponRadio infrastructure is almost in place for the broadcast industry, represented by these potential advertisers.

Attached to the printer is a CouponRadio terminal box, specially designed to process the infomessages and other data stored on the RadioCard. This terminal box is designed to work in cooperation with the creditcard verification terminal, already in place, and in no way will it effect its service. The terminals just share the same printer.

```
MANHATTAN MOTORS, INC.
500 W 59TH ST.
NEW YORK, NEW YORK 10018
212-582-7700
4/30/93                1:03PM
INFOMESSAGE:
4/10/93 WQHT 2:15PM
MANHATTAN MOTORS
500 W 59TH ST.
212-582-7700
$100 OFF BEST PRICE
I THE RADIOCARD HOLDER
RECEIVED THE PROMOTION
OFFERED ON THE RADIO.
John Q. Citizen
```

Figure 3: Sample Coupon

When a customer's RadioCard is inserted into the advertiser's terminal box, the unit is programmed to instantly search the card for that advertiser's specific coupon formatted infomessage, retrieved earlier from the radio broadcast. **Fig 3.** Once found, the special terminal both prompts the printer to print a RadioCoupon, and then erases the infomessage from the RadioCard, leaving all other infomessages intact.

It is proposed, that the CouponRadio terminal boxes be provided to the advertiser, by the broadcaster, for a specified amount of time, depending upon the run of the advertisement. Printers would also be available to the advertiser in the same fashion. It would be entirely the radio station's decision, depending upon the advertising package arranged, as to the nominal cost charged to the advertiser for the use of the CouponRadio terminal box, printer or both.

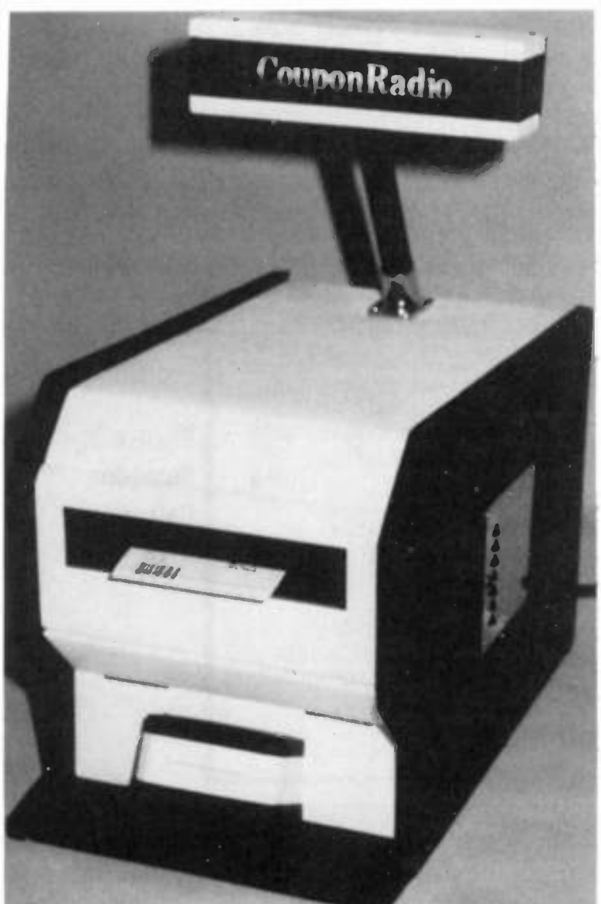


Figure 4: CouponRadio Portable Kiosk Demonstrator

This CouponRadio System would have the powerful effect of a listener being led directly to an advertiser offering a promotion, and then cashing in on the offer with a merchant-generated RadioCoupon, showing proof of listening. To this authors knowledge, there is no other advertising medium to date that would be able to compete with this kind of advertising focus - audio, visual, interaction, and printed coupons in one medium.

Figure 5: Sample Kiosk Coupons with Graphics

A second CouponRadio printing system makes use of a heavy duty, thermal, kiosk type printer, installed in attractive kiosks designed for unattended, self service operation. **Fig. 4.** These kiosk units are equipped with a computer monitor which will prompt RadioCard holders to insert their RadioCard and instantly receive all their infomessages separately printed in the CouponRadio coupon format. **Fig. 5.** The computerized kiosk will automatically erase each infomessage from the RadioCard as the RadioCoupon is being printed. At this time, it is proposed that these kiosk printed coupons will be accepted by the advertisers whose infomessage appears on the RadioCoupon, even though the advertiser did not print the coupon. If an advertiser desires, the broadcaster can attach a special code to the infomessage, which will also cause a UPC bar code to be printed on the RadioCoupon.

When the RadioCoupon printing process is completed, RadioCard holders will be prompted to remove their reusable RadioCard from the kiosk. In addition, after the RadioCoupons are printed, each kiosk printer unit can be programmed to automatically provide the RadioCard holder with other valuable in store coupons. Both the RadioCoupons and the in-store coupons may contain a corporate or food franchise logo, promoting the sponsors.

The kiosk units are initially planned for use in record store chains. However, at this time, other giant retail chains are expressing interest in the CouponRadio kiosk units, as the acceptance of CouponRadio grows. The CouponRadio kiosks are unique from other kiosks in retail establishments where an individual must stand in front of the computer screen and answer numerous questions in order to get a little information. The CouponRadio kiosks are designed to print in coupon format information already provided by the broadcaster. Listener interaction with the CouponRadio kiosk will be quick and fun.

For coupon security, as well as to encourage direct interaction with advertisers, there are no current plans for RadioCoupons to be printed at a listener's home. Although provided for in the earliest of the CouponRadio patents, it was additionally decided against having RadioCoupons printed in automobiles or radio receivers. Studies showed that listeners would never replace the printer paper, vehicles were too hard of an environment on printers, listeners were afraid of having to take their cars to Epson or Hewlett Packard to have the printers fixed, and the RadioCard was sexier and more advanced technology.

Implementation

Corporations doing business in today's complex and global environment, understand how vital it has become to form mutually beneficial alliances with other companies, exchanging specialized technology and ideas necessary in bringing a product successfully to market.

A SYSTEM approach to CouponRadio has been achieved with the cooperative efforts of RCS Corp.; an international industry leader in music scheduling software for radio; Rohde & Schwarz, Inc.; an international, high tech manufacturer of RBDS encoders and other specialty broadcast and test equipment; and industry veteran, David Reeves; Chief Engineer of WHZT-Z100 FM in New York; all dedicated to broadcast excellence.

In order to maximize the efficiency of the RBDS encoder, it should be understood exactly how much data the encoder is capable of transmitting, and how fast. There are approximately 11 different groups that an RBDS encoder can transmit, each providing a different service. For instance, Group 0A = 2 characters of a stations call letters w/ 2 alternate frequencies, Group 2A = 4 characters of radio text (infomessage), Group 4A = clock time/date, Group 15A = 4 characters of a stations call letters w/o alternate frequency, etc. Every 2 seconds the RBDS encoder is capable of transmitting 23 of these groups, known as a RECORD. It is very important to have a plan of what you are trying to accomplish with the data encoder, since it is not possible to transmit a record containing many **different** groups, and still have all the features of the groups reach the radio receiver, in their entirety, in the same amount of time. If a broadcaster makes a decision to transmit infomessages, he must also decide whether it will take 2 seconds to transmit a 64 character infomessage, or 32 seconds. Once you have a goal, you can program the encoder with the appropriate blend of groups that fits your needs. The United States RBDS Standard, page 13, table 4 shows graphically how these different groups relate to each other within the 2 second time frame. Due to typical broadcast transmission problems or a listener tuning to a station in the middle of a broadcast, a broadcaster should usually assume that it will take twice as long then calculated for many of the different RBDS features to reach and update a radio receiver. This is one of the reasons why the standard provides for different groups to repeat themselves many times within the 23 group-2 second time frame.

During our CouponRadio tests and demonstrations our plan was to transmit a record consisting of: 20- 2A groups, 2- 15A groups and 1- 4A group. This assured us that our infomessage, our call letters, the date and time would reach our receiver in less then 2 seconds. The inclusion in our record of other groups with different RBDS features would have slowed down the transmission of our infomessage. It was felt that when an advertisement was being broadcast, the advertiser was paying for the infomessage to be transmitted with no time wasted. When other more lengthy programing is being aired, a broadcaster may switch his encoder to broadcast a different blend of groups.

Transmitting infomessages should be an always or never decision. If a broadcaster transmits an infomessage once, listeners will depend upon them always. If there is no dependable routine for a listener to rely on, they will be uncomfortable with the service. The success of CouponRadio depends upon the broadcaster effectively transmitting changes of valuable infomessages, precisely synchronized to the corresponding, regularly scheduled program material. Because sudden and unexpected program changes by a program director are normal, MASTER CONTROL, supplied by RCS, Corp., was used to achieve the ultimate synchronization of the radio infomessages, and to facilitate non-stop, hands-off testing of the CouponRadio System.

Master Control is a complete integrated studio system, providing state of the art digital audio. Master Control was interfaced to the Pioneer 300 CD juke box, to play CD's without jock intervention. Master control also controls the stations audio, cueing and firing of CD's, digital carts and other digital sources, while sending the corresponding and precisely synchronized infomessages, from the SELECTOR database, to the RBDS encoder exactly at the correct time, all under the jock's control. Master Control runs on a Novell network, and is completely network aware, which means last minute changes can be incorporated easily and transparently.

For demonstration purposes, Master Control was connected directly to the RS-232 port of the RBDS encoder. Under normal circumstances, it is practical to have a PC computer as an intermediary which provides immediate access to the full capabilities of the RBDS encoder. To facilitate direct communication with the encoder, RCS Corp. prefaced the infomessages with a special command which directed the data into the proper RBDS group.

CouponRadio, Inc. is currently working on other systems which will help to automate and synchronize the transmission of infomessages with existing broadcaster equipment, such as CD players and 'Carts.'

Provided in the Appendix, are flow chart diagrams which should give station engineers suggested preparation of broadcast equipment so that the automatic transmission of infomessages can become an almost invisible and routine operation of daily broadcasting.

Rationale

From a business and marketing standpoint, the ability to display one's name, slogan, call letters, format, special number etc., is not new. For years, the Department of Motor Vehicles has been in that lucrative business by manufacturing personalized license plates, better known as vanity plates, to customers wanting to display their message. Similar to the Electronics Industry, (manufacturers of radios that provide a similar service for broadcasters), the Department of Motor Vehicles offers up to 8 character spaces, in which to display one's message. From a profit standpoint, although it is nice to post your message, the only group that makes any documentable income from this type of promotional medium, are the manufacturers who actually provide the hardware.

Today, in order to go into a new and profitable business venture, one should have a business plan. RBDS is a brilliant technology without a plan, at least not for the broadcasters. In order for it to have any financial impact, it needs to be embraced in a business plan. By itself, RBDS cannot generate the revenue broadcasters would like, if at all. Except for the paging industry, any RBDS proponents claiming profits for broadcasters, most likely manufacture the hardware, and would like sales today.

It is this author's opinion, as well as others, that CouponRadio may be the business plan broadcasters are looking for, and will consider. The CouponRadio plan has been prepared and updated since approximately 1989. The plan offers broadcasters a new way to compete, as a group, against other media looking to steal listeners and advertisers away.

Periodically, the newspapers and magazines report on new interactive or service oriented media, or products, designed to gain more of the attention of the listening public, a group once owned by radio. The automobile, a listening environment, at one time exclusively belonging to the radio, has had to make room for the cassette, CD, mobile phone etc. Even while walking, jogging or at the beach, these new electronic gizmos now go along.

To effectively compete during these new electronic times, it is felt that CouponRadio can be the new, highly profitable approach broadcasters are looking for. The broadcaster's effective use of infomessages, and the necessity of the listener to have the ability to save them, will give the listener better service and promote listener interaction. Higher broadcaster revenue should come from the additional demand and/or cost, to advertise on this new and exciting medium. It will take more then just displaying a station's call letters or music format on the radio to increase profits and win back advertisers. By combining those efforts together with the CouponRadio plan, radio broadcast should re-establish itself as the premier advertising medium, a listener's source for endless advertiser promotions and giveaways. Unlike other media, a radio broadcast is free to the listener. The CouponRadio plan is not to inhibit advertisements, but instead to make them great and memorable.

The CouponRadio plan calls for a planned introduction, similar to that of the cable industry, where broadcasters in the same area can gear up together, fairly, and listeners can be told when they can expect the new service. The lack of a plan may cause an unnecessary run on encoders, as broadcasters compete to keep up with each other. Las Vegas, Nevada, is a perfect example of a city where the broadcasters have started transmitting RBDS together. It represents the benefits of a planned and orderly introduction of a new technology.

Beneficiaries

Who benefits from CouponRadio is as simple as **A, B, C. Advertisers, Broadcasters, Consumers,** and many others.

Advertisers of all sizes will quickly realize that radio is not the same advertising medium it always has been. It is new, interactive and unforgettable. Advertisers will want to rethink their marketing strategies to take advantage of this new medium which includes the use of infomessages and the powerful result of a consumer's ability to save them. The advertiser's ability to turn the infomessage saved by a listener into a promotional coupon, by the use of a CouponRadio printer, should become a very powerful marketing tool, literally putting a coupon in the hands of those listeners who want it.

Advertising agencies and marketing groups will no doubt be called upon by their clients for their expertise in designing new and effective advertisements and promotional giveaways that maximize the full potential of radio's new and dynamic capabilities. Agencies will realize that the potential, and the results, of advertising on radio is altogether new and must not be taken for granted. Radio will be seen as having the capabilities, currently unavailable to any other advertising medium, and therefore must be included in most marketing budgets.

Broadcasters should welcome, after periods of declining revenue, the attention and interest their new advertising and service oriented medium is getting from others. The new ability of making what is heard on the radio **unforgettable**, should make history repeat itself by making radio, once again, the #1 marketing and promotional tool for the advertising industry. Since the new benefits of advertising on radio will be obvious to most, radio station account executives will spend less time on selling radio as a medium, and more time on taking new and creative advertising orders. CouponRadio will give new meaning to interactive promotional games and contests sponsored by the broadcaster. By encouraging listeners to send in their proof-of-listening coupon, broadcasters can review the statistical data on the coupon to determine listening habits.

Consumers, the listeners of radio, will appreciate radio becoming a more service oriented, and interactive medium. By having the ability to convert infomessages into promotional coupons, listeners will be able to take advantage of the substantial savings offered by promoters, ie. advertisers, record companies, broadcasters, etc. Studies have shown that consumers are loyal and respond favorably to those who provide them with good service. Listeners have, for a long time, expressed frustration in not being told the name of a song, or having missed an important telephone number. Now, with the ability to save an infomessage, listeners will see radio as a full service medium. As the population becomes more diverse, not all listeners are literate in English. With CouponRadio, a Hispanic, Asian or other individual listening to an English speaking station, can overcome the language barrier at the push of a button. They can take advantage of promotions by printing a RadioCoupon.

The record industry, equally as frustrated, have continuously asked broadcasters to give listeners the song title and artist information when music is being played; "play it, say it." This information is crucial to new song promotion and sales. In the form of infomessages, the song title, artist, record label, and even catalogue number, can be instantly transmitted to a radio, whereby a listener can "hear it, save it." By taking their RadioCard to a participating record store, with a CouponRadio kiosk, listeners can have printed out in coupon form, the titles of the music they wish to own. As soon as a song is broadcast for the first time, record sales can be realized almost instantly by this process. It is the author's experience that often, by the time I learn the name of the song, I become tired of it. Many record sales are probably lost in this manner. In one situation, I accidently scanned my radio to a Spanish speaking station. I heard the hottest piece of music. I had to have it. Unfortunately, I missed the title because I don't speak Spanish. If I had a CouponRadio receiver, all I would have had to do was push the memory button.

Record stores equipped with the CouponRadio kiosk, can further program the unit to print additional in-store coupons that may provide the consumer with additional savings. To further promote the use of the CouponRadio kiosk, record stores can program the unit to count the number of customers, awarding the "magic number" customer with a big payoff of cash and prizes, cooperatively sponsored. CouponRadio could be just what the record industry needs to ease pressure on broadcasters, and increase sales.

Recording artists, who make much of their income from royalties from the sale of their music, would benefit from CouponRadio, and most likely, publicly endorse the system. Many of these celebrities set the trends that the public follows. Recording artists can directly benefit financially from CouponRadio, as well as help to make the market.

Handicapped listeners, a group not initially thought of when discussing radio, would benefit greatly from a radio broadcasting system like CouponRadio. Since this author's father is handicapped, I know first hand how frustrating it must be for a disabled person to search unsuccessfully for a pencil to write down an important telephone number or message. In cooperation with the electronics manufacturers, it is proposed that in addition to the standard memory button, radios can be outfitted with a receptacle, to which special remote memory button devices, designed and manufactured to suit the listeners capabilities, would plug into. When the listener desires to save an infomessage, all that is necessary is to activate his special memory button device. The infomessage is then stored on the RadioCard, and available later for recall by the listener, or an assistant. In this situation, implementation of the CouponRadio plan may be applauded by the media and special interest groups, as responsible broadcasting and electronic equipment manufacturing. It may turn out that the handicapped provide a substantial and immediate amount of radio receiver sales, since the many disability programs that are available, may end up paying for the units anyway. It will be a rewarding feeling for the broadcast industry to know, that the handicapped can also take advantage of promotions heard on the radio.

Electronics manufacturers, may soon realize the long term effect of launching the CouponRadio plan, as compared to the short term effect of the RBDS receivers currently being introduced. More then just a radio receiver that displays call letters, the CouponRadio receivers open up such a world of possibilities, that it may be necessary to manufacture most radio receivers (walk types, auto, portable and home) with the CouponRadio features. By participating in a multi-industry launch of CouponRadio, as opposed to just an electronic industry launch of RBDS, radio receiver sales should be immediate by listeners who do not want to be left out of the interaction. It is human nature to participate in new trends and not to be left out.

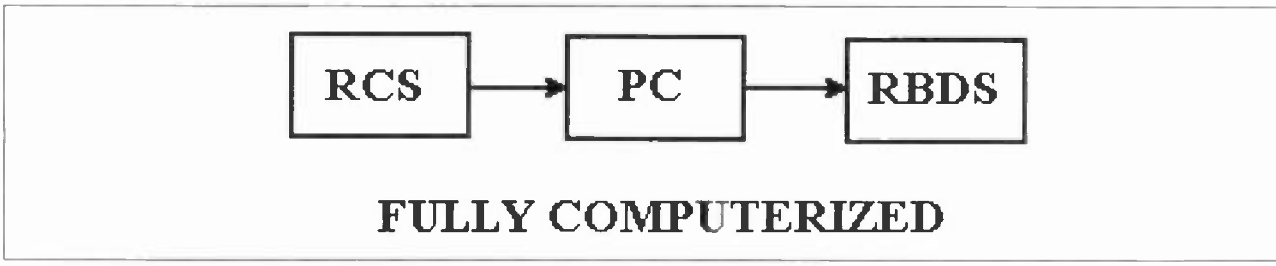
All groups will certainly benefit from the introduction of the RadioCard. Looking exactly like a creditcard, it should prove to be a new multi-industry billboard, on which broadcasters, electronics manufacturers, automobile manufacturers, record companies, record stores, and other promoters will place their colorful and exciting logos. These RadioCards should become an instant hit with the listening public, and may possibly become the ultimate collectible. Savvy marketers may introduce desirable, limited edition, RadioCards to perpetuate the excitement and their value. RadioCards will be included with the purchase of a new radio receiver, sold as replacement cards in electronic and record stores, or given away as a promotion device. RadioCards will show others the brand radio you own, the automobile you drive, the record store you shop at, or that you were fortunate enough to get the promotional piece from your favorite radio station.

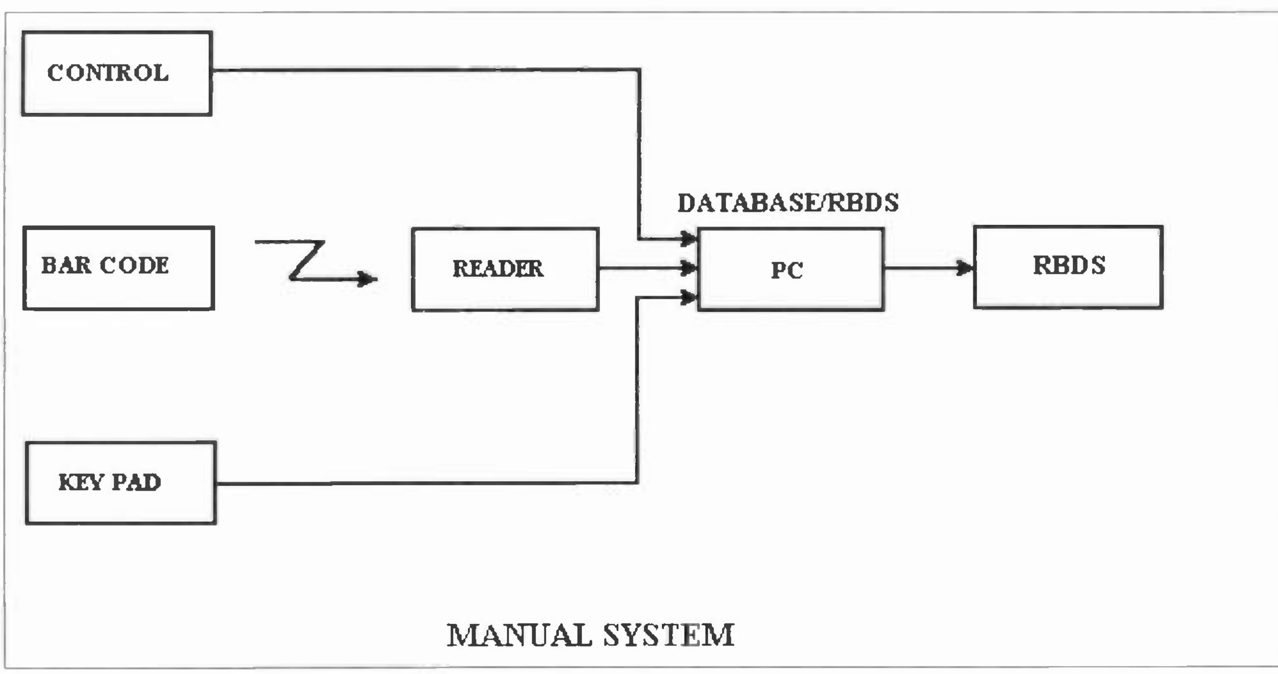
Conclusion

If the CouponRadio plan is accepted by the broadcasters, then more work is needed to be done. Broadcasters may want to learn more about using the RBDS standard, investigate the ways available to automate the transmission of infomessages, but most importantly broadcasters must request that more radio receivers be equipped with CouponRadio features. To this author's knowledge, it has always been that radio receivers have been built based upon the signal the broadcaster was transmitting. Unlike today, where a broadcaster is being asked to transmit a signal based upon receivers that are already built. Radio receivers need to be designed and manufactured so that broadcasters can profit from the signal they transmit. If broadcaster benefits and profits were the electronics industries true goal, and selling point of RBDS, then after reviewing the CouponRadio plan, it should be clear that the electronics industry needs a new hit, and the radio industry needs a change. The radio industry needs CouponRadio. To quote Ross Perot: "If there's a better idea, I'm all ears!"

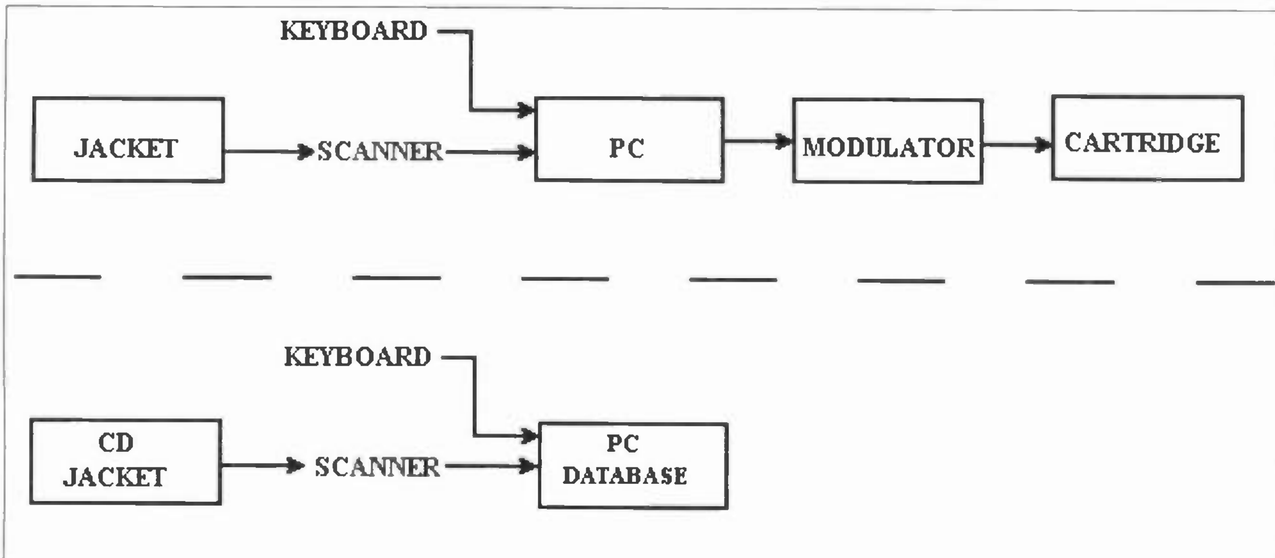
Note: Sample radio station logos used in fig. 5, were used only to show how the proposed kiosk type RadioCoupons will look. The use of the sample logos, in no way imply the stations involvement or endorsement of the CouponRadio project at this time.

APPENDIX

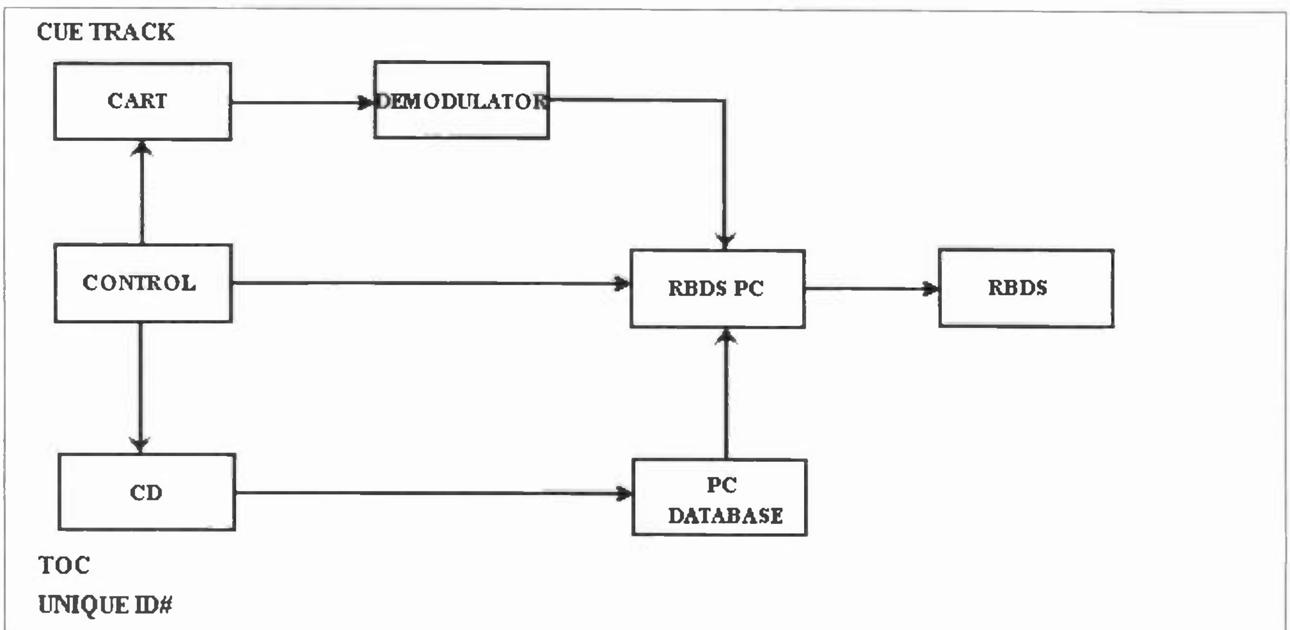

RCS
PC
RBDS
FULLY COMPUTERIZED


CONTROL
BAR CODE
READER
DATABASE/RBDS
PC
RBDS
KEY PAD
MANUAL SYSTEM

APPENDIX



SET UP / SEMI AUTOMATIC



SEMI-AUTOMATIC

RDS/RBDS INSTALLATION GUIDELINES AND FUTURE INTERFACE APPLICATIONS

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RE America, Inc.
Westlake, Ohio

ABSTRACT

On January 8, 1993, the United States Radio Broadcast Data System (RBDS) Standard was officially released by the National Radio Systems Committee. As this new era in datacasting begins, FM broadcasters are faced with integrating RDS technology within their present transmission scheme. This paper is intended to serve as a reference and offer guidelines to aid broadcast engineers in the installation of the RDS subcarrier system. The acronym RBDS, is associated with the title of the U. S. standard document which is based on the European Radio Data System or RDS technology. For the benefit of this paper, we will hereon refer to the technology simply as RDS.

TECHNOLOGY BACKGROUND

RDS is a subcarrier data transmission system that was developed by the European Broadcast Union which is added to an FM composite signal at 57kHz. It requires a very low injection level due to its robust nature. Being the third harmonic of the pilot frequency, it is phase locked to 19kHz. RDS is a double side band suppressed carrier that is amplitude modulated by a shaped and biphase coded data signal. The basic data rate of the system is approximately 1200 baud (1187.5bps).

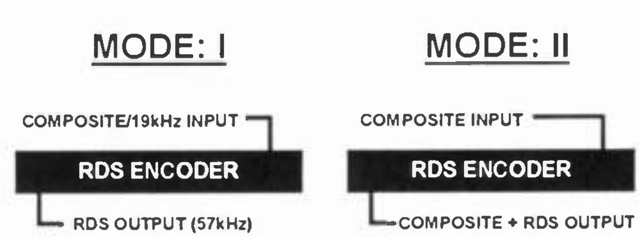
INTRODUCTION

A vast majority of FM broadcast facilities today have invested in a variety of audio signal processing devices that are recognized for their ability to provide an FM station with a unique and competitive sound. Though audio processing may offer signature qualities to the air product, care must be taken when adding subcarriers such as RDS into a processed composite signal chain. As a general rule in maintaining bandwidth and spectral integrity, subcarriers should be added post audio processing. While there may be certain instances that involve filters and processing which allow subcarriers to pass through unaffected, it is considered good practice to make subcarrier injection the final component in the composite signal chain.

The following procedures have been assembled in a hierarchy format to guide the broadcaster in the decision making process when installing an RDS encoder.

1. DETERMINE SIGNAL PATH CONFIGURATION

Many RDS encoders are capable of functioning in one of two modes of operation which are either as of a stand-alone subcarrier generator source or as an adder, where the composite signal and the RDS subcarrier are combined. This allows the RDS encoder to be inserted into the composite signal chain. Based on these choices of mode operation, you may first decide the best physical placement of the RDS encoder.



(Fig. 1)

MODE I: The RDS encoder is operating as a subcarrier generator source. (The composite signal provides a source for phase lock only.)

MODE II: The RDS encoder is operating as an adder that combines the RDS/subcarrier with the composite signal.

2. DETERMINE INSTALLATION SITE

Once the RDS encoder mode choice has been made, we may proceed to the next step which is the location of the installation site. Ideally, locating the RDS encoder at the studio is desirable due to the ease of interfacing with a PC for programming. In this case, the data can be conveniently downloaded through the RS-232 interface located on the RDS encoder.

Broadcasters who have the studio and transmitter located at the same site should have a straight forward installation. When this is not the case, other factors that may preclude the RDS encoder installation at the studio such as;

- Limited full-time auxiliary input capacity on a STL.
- No available auxiliary input capacity on the STL.
- Bandwidth limited input capacity on the STL.
- Discrete audio without composite spectrum capacity on the STL.

If any of these conditions exist at the studio site, you will most likely need to install the encoder at the transmitter site. While this may seem inconvenient from a communications standpoint, it should be noted that in many cases there are existing remote control facilities with spare capacity that may be utilized. Methods of data communication to the RDS encoder are discussed in section 6 of this paper.

3. RDS SUBCARRIER INJECTION LEVEL (MODES I&II)

Regardless of the choice of mode operation (MODE I or II), the RDS output level of the encoder must be calibrated to the input sensitivity of the next channel input of the signal chain. By referencing the technical manual of the STL, exciter or transmitter, a specification of input sensitivity in volts peak/frequency deviation can be found. The desired RDS injection level in terms of frequency deviation is then multiplied by this ratio to determine the output level required of the RDS encoder.

EXAMPLE:
Exciter input sensitivity is 1.5Vp/7.5kHz (10%)
Desired RDS injection level is 2kHz (2.7%)
Required RDS encoder output is:

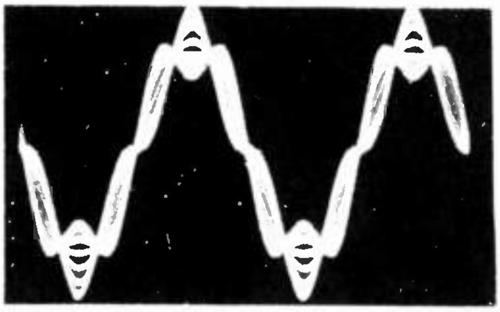
$$\frac{2\text{kHz} \times 1.5\text{Vp}}{7.5\text{kHz}} = 400\text{mVp}$$

The RDS encoder output may then be adjusted utilizing an oscilloscope or peak reading voltmeter to this output level.

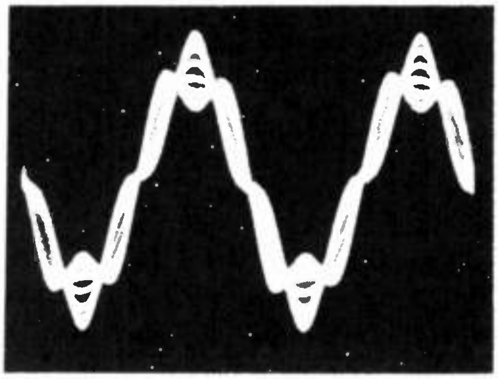
5. RDS PHASE ADJUSTMENT

After injection levels have been completed, the phase relationship of the 57kHz subcarrier and the 19kHz pilot is the next step in the installation process. This is an important adjustment and should not to be overlooked. It is possible for a mis-adjustment to cause low level interference in the main channel as a result of intermodulation products. Proper synchronization and phase adjustment to the pilot, such that the zero crossings coincide, will effectively eliminate the problem.

The adjustment is performed by referencing the composite signal from an off-air monitor with an oscilloscope. This can be done by adjusting the phase control on the RDS encoder for the proper relationship to within ±10°. The waveform in Fig. 2A illustrates correct phase relationship of 19kHz and 57kHz while Fig. 2B illustrates a phase-error of 10°. Dedicated RDS monitors are also available with the ability to conveniently display the phase relationship of RDS and the pilot signal.



(Fig. 2A)



(Fig. 2B)

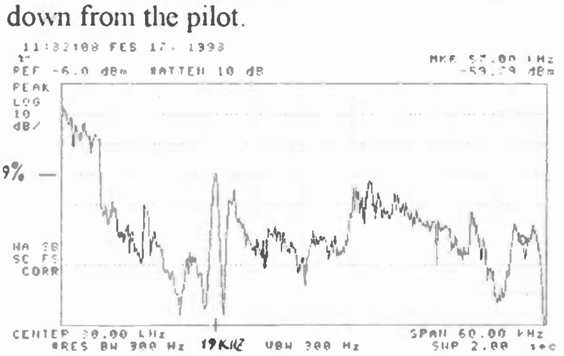
4. MONITORING RDS INJECTION LEVEL

Monitoring the RDS injection level can be done through the use of a spectrum analyzer capable of low frequency operation to examine the composite signal from an off-air monitor. (Fig. 3)

MONITORING RDS

(Fig. 3)

Before connecting the RDS encoder to the signal chain, it is recommended to have the output level of the RDS encoder set to minimum. This will prevent any accidental overmodulation from occurring. While monitoring the composite signal with a spectrum analyzer and referencing the pilot, increase the RDS encoder output level until you are at the desired injection level (Fig. 4). If the spectrum analyzer is efficient enough to resolve the RDS modulation sidebands, remember to subtract 6db (sidebands added together equal total RDS subcarrier modulation). In other words, if the RDS subcarrier is injected at 2.6% and the pilot is at 9%, then the RDS signal will be 10.8db down from the pilot. As a result, individual sidebands are now 16.8db down from the pilot.



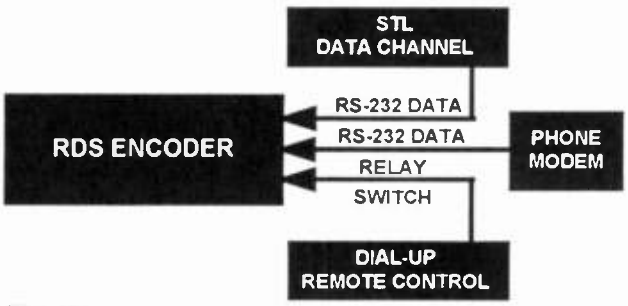
(Fig. 4)

6. RDS DATA INTERFACE

Once an RDS encoder has been installed and properly added with the FM composite signal, there is a need to interface and communicate the dynamic RDS data that is being broadcast. True, RDS data could be fixed or static and never require any updating however, in the RDS world it would be much like playing the same audio program continuously. This section will briefly explore two known methods of routing data for control of the RDS encoder and what immediate future lies ahead for dynamic data control.

DATA CONTROL METHODS (Fig. 5)

- Predefined data that is stored internally to some encoders may be selected via a relay closure switching matrix. This could include interfacing with an existing dial-up remote switching device.
- Dynamic or real-time control through a phone modem to the RS-232 port on encoder.
- Dynamic or real-time control through a data channel on an STL to the RS-232 port on encoder.



(Fig. 5)

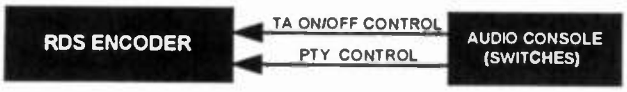
RDS DATA CONTROL METHOD I:

Data control (Method I) may utilize an encoder that offers some internal RDS data storage. In this case, you may connect switching devices to a relay closure pinout that is located on the encoder. This particular switching scheme could work with an existing dial-up remote located at the transmitter site (Fig. 6). An encoder that is located at the studio could interface through a direct wired system. One possible switching scheme could make use of the fader or mute buttons located on the air audio console for control of Traffic Alerts (TA) or format (PTY) changes (Fig. 7).

TRANSMITTER SITE DATA CONTROL

(Fig. 6)

STUDIO SITE DATA CONTROL

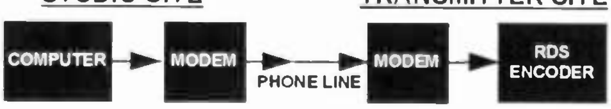


(Fig. 7)

DATA CONTROL METHOD II:

Dynamic or real-time data control will require a PC, a control program and an interface to the encoder via the RS-232 port. Much like any computer peripheral device, the RDS encoder must have the correct communication parameter set-up such as; baud rate, stop bits and parity. Once these fundamental communication requirements have been established, use of an off-the-shelf phone modem on standard phone lines could serve as the data communication interface between the studio and the transmitter as shown in the block diagram (Fig. 7).

STUDIO SITE TRANSMITTER SITE



(Fig. 7)

CONCLUSION

As the evolution of RDS progresses, dynamic datacasting will give program directors and air personalities an entirely new way of communicating with their listening audience. By sending text messages such as song titles and artist labels, it is envisioned that RDS radio text will soon provide broadcasters with an added revenue source from many advertisers. Additional revenues through the lease of private data channels available on RDS such as Radio Paging (RP) and Transparent Data Channel (TDC) as their demand increases through the coming years.

The future of RDS data control will grow rapidly from operator assisted entries to fully automated RDS data control. Currently, a variety of software programs exist for playlist and traffic programming. As of this printing, an RDS window or menu is already in the making from these software suppliers.

Another future interface will involve the alerting features of RDS which are being considered by the FCC for the Emergency Broadcast System. As a major improvement over the current EBS program, RDS alerts can be delivered instantaneously to all participating stations. Whether the alert is given by a switching relay or through a coded PC program, the American public will feel a greater sense of security through the use of RDS "smart" radios in the nineties. Radio broadcasters will benefit from a much needed technology that will offer the industry a new level of FM broadcasting through the enhanced information delivery system found in RDS.

ACKNOWLEDGMENTS

We wish to express our extreme gratitude to the people and organizations responsible in the adoption process of the RDS/RBDS technology standards. Our additional thanks to the staff of RE Technology AS and RE America, Inc. for their unending effort and development in the fields of broadcast, communication, test and measurement.

1.) United States RBDS Standard, January 8, 1993, NRSC Document, ©1992 EIA/NAB.

2.) EBU Cenelec Standard EN50067, 1984, 1986, 1990, 1991, 1992.

3) FCC FO Docket 91-301, FCC FO Docket 91-171 and FCC 92-439 Notice of Proposed Rule Making/Further Notice of Proposed Rule Making

4) RE #983-408, RE533 Operating Manual.

5) RE #983-302, RE531 Operating Manual.

6) RE #983-405, RE331 Operating Manual

NHK'S HIGH CAPACITY FM SUBCARRIER SYSTEM

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NHK has developed an FM multiplex broadcasting system using a new digital modulation method called L-MSK which is a kind of MSK. This system is multipath-proof, and has a transmission capacity of 16kbps. Moreover, it has a compatibility with RDS because they use different frequency bands. Transmission tests were carried out to measure the performance of the 16kbps L-MSK system in the service area of the NHK Tokyo FM station. These tests revealed that this system was superior to RDS in terms of the correct reception rates obtained in mobility. FM multiplex broadcasting using L-MSK has many merits for traffic information services.

1. INTRODUCTION

Most big cities in the world suffer chronic traffic congestion. Car drivers want real time traffic information to avoid heavily congested roads. A new traffic information service using FM multiplex broadcasting is in urgent need. The FM transmission system, RDS, described in CCIR Recommendation 450 is adopted in all regions (region 1:Europe and Africa, region 2:America, region 3:Asia). In Europe, RDS has been put into operation. The RDS system has a transmission capacity of about 1.2kbps. NHK has developed a new way to use MSK in a FM subcarrier application, called L-MSK. It is multipath-proof, and has a transmission capacity of 16kbps. This system and RDS systems are compatible with each other because they use different frequency bands.

FM multiplex broadcasting can provide new services such as character and graphic information by means of multiplex digital signals with existing FM broadcasting signals. Since transmitters and antennas which have been installed for FM broadcasting can be used without any modification and its service areas are very wide, FM multiplex broadcasting can be introduced at a low cost.

This paper describes the characteristics of FM multiplex broadcasting paths, the outline of a new digital modulation method called L-MSK which has been developed for mobile reception, the result of transmission tests and its applications to the traffic information services.

2. FM MULTIPLEX BROADCASTING PATHS

In FM multiplex broadcasting, digital signals are multiplexed with the stereo composite signals by frequency division multiplexing. The multiplex signals should be multiplexed between 53kHz and 100kHz. And the multiplexing level is to be kept as low as possible for the compatibility with existing stereo sound.

Generally, in the digital transmission for mobile reception, multipath interference is a serious problem. In cases where there is multipath interference in receiving FM multiplex broadcasting signals, the resultant inter-modulation deteriorates S/N of the stereo sound signals. Conversely, interference of the stereo sound signals with the multiplexed digital signals deteriorates digital transmission characteristics. Disturbance of the multiplexed digital signals by the stereo sound signals affects nearly the whole multiplexable frequency band.

3. DIGITAL MODULATION METHOD AND ERROR CORRECTING CODE

3.1 L-MSK

Fig.1 shows the baseband spectrum of FM multiplex broadcasting signal. The FM multiplex digital transmission characteristics affected by the multipath interference depend on the modulation level of the stereo sound signals. Errors do not occur at a multiplex level of 2.5% if the stereo sound signals are not modulated, however acceptable reception is not possible unless the multiplex level is approximately 10% when the signals are modulated. A system to control the multiplex level of digital signals by the modulation level of the stereo sound signals can be considered. The multiplexing level of the digital signals is controlled by the L-R signal level. MSK with an amplitude level which is controlled as mentioned above is referred to as L-MSK (Level controlled Minimum Shift Keying). Fig.2 shows a waveform of the stereo sound signals and a modulation signal waveform of L-MSK.

By using L-MSK, which increases the multiplexing level only during an increase of the modulation level of the stereo sound signals, the digital transmission with the best energy efficiency under multipath interferences is possible. To ensure the compatibility problem also, the multiplexing level should be kept low if the stereo sound modulation level is low.

3.2 Comparison of digital modulation methods

Fig.3 shows bit error rates measured in field tests changing the bit rates - 9.5kHz and 19 kHz - and the modulation method. The bit error rates of RDS also have been measured for comparison. Two-channel space diversity was used for these tests. Three typical roads, - ordinary, multipath and highway -, were selected. Considering the recent progress in error correcting code techniques, a bit error rate before error correction of a service boundary is about 10^{-2}. Fig.3 shows that L-MSK has the best performance among other digital modulation methods, there is a possibility of 9.5kbps or 19kbps, and the performance of L-MSK is almost equal to that of RDS.

3.3 Error correcting codes

Computer simulations on some kinds of error correcting code have been carried out using error pattern data gathered in field tests. As a result, it has been revealed that the product code of the (272,190)code is the best. Fig.4 shows the frame structure of FM multiplex broadcasting. The error correction capability of the (272,190)code is very powerful and some kinds of decoding LSI are already on the market. The (272,190)code is adopted for other digital systems, for example, protecting key data of pay TV, teletext, facsimile broadcasting, FM multiplex broadcasting for stationary reception, optical memory card systems, etc..

4.TRANSMISSION TEST

Transmission tests were carried out to measure the performance of the 16kbps L-MSK system in the service area of the NHK Tokyo FM station. Table 1 shows the transmission parameters of the test. All highways and main national roads in the area were selected for the tests, and the measured distance totaled more than 1500km. Transmission data could be received without error during the 90% of all measured time by carrying out error correction with the product code of (272,190) code as previously mentioned.

Cars are subjected to intense multipath interference or cut-off in tunnels, therefore 100% correct reception at all places is impossible. It is necessary to transmit same data several times. The received traffic information is displayed on a LCD or CRT display page by page. Correct reception rate is defined as below.

$$\text{Correct reception rate} = \frac{\text{Number of received correct pages}}{\text{Number of all transmitted pages}}$$

Fig.5 shows the iterative transmission effects. This figure shows the proportion of files obtained with more than 95% correct reception rate in the FM service area. One file contains measured data in eight minutes. Assuming that the limit for broadcasting service is at least 95 % in correction reception rate, the service is available to more than 90 % of the relevant FM service areas even with the packets as many as 250 (1 packet = 22 bytes) by means of three-time iterative reception.

Fig.6 shows the roads where more than 95% correct reception rates can be obtained assuming one page to be 15packets in the FM service area of the NHK Tokyo FM station. A correct reception rate of more than 95% can be secured with one reception within approximately 20km of Tokyo Tower where the transmission antenna is installed. In case of three time iterative transmission, more than 95% correct reception rates can be obtained in almost all service areas.

Fig.7 show the correct reception rates of L-MSK in comparison with that of RDS in several measuring areas. Both are the correct reception rates received in mobility by transmitting files of approximately 350bytes. Since the error correction code used in L-MSK is powerful, L-MSK is superior to RDS.

5.SYSTEM PROTOCOL

The Electrical Telecommunication Technology Council of Ministry of Posts and Telecommunications is deliberating on the system standard of the new FM multiplex broadcasting system in Japan. The system standard will be described following to the 7-layer reference model for data broadcasting that is shown on CCIR Recommendation 807. Fig.8 shows an outline for the system protocol of the new FM multiplex broadcasting system in Japan.

6.APPLICATION TO TRAFFIC INFORMATION

6.1 Service areas of FM multiplex broadcasting

FM broadcasting covers 99.9% of Japan and 100% in areas where traffic information is needed. Therefore all that FM broadcasting stations should do is to broadcast detailed traffic information in their service areas and around them. Drivers can tune into them.

In cases where plenty of transmission capacity can be used, however the stations should broadcast traffic information including that of other service areas to make it more usable for drivers.

6.2 Bit rate for traffic information services

Traffic information services will be gradually provided after the development of infrastructure such as detection equipment for the speed of cars running on roads. Table 2 shows a service model divided into three levels. It is assumed that the services area is Tokyo, where traffic congestion is the severest, 30% of all roads are jammed, and the same traffic information is sent twice every five minutes.

Even at the final stage of level 3, a transmission capacity of about 5kbps is enough to broadcast sufficient traffic information. Therefore, the remaining can be used for services other than traffic information, such as news, weather forecasts, stock market information and paging. Naturally, sound information service can also be provided using the state of art bit reduction techniques for digital sound signals.

6.3 Service images

In regard to proper data broadcasting with high reliability for car drivers, FM multiplex broadcasting using L-MSK is the first example. This system can send real time traffic information and display it on a road map display of navigation equipment or on a special LCD display and can also output synthesizer sound. It is possible to provide traffic information which car drivers can easily comprehend.

Besides traffic information, characters and graphics containing downtown information such as department stores and movie theaters around a station can be provided. For stationary receivers, this system can also broadcast character and graphic information related to the stereo program, for example names of its composer and player, and song' texts.

7.CONCLUSIONS

In Tokyo, it is said that traffic congestion will be avoided if 3% of car drivers on ordinary roads and 13% on highways have traffic information to keep away from such congestion. The early introductions of rapid and accurate traffic information services are strongly required for car drivers. There is a new movement to combine some systems that have been individually developed so far, such as IVHS in the United States and VICS in Japan.

FM multiplex broadcasting has many merits for traffic information services. It needs no new radio waves, and offers wide service areas, low running costs, and complete networks. It is expected that FM multiplex broadcasting play a vital role in the traffic information service for car drivers.

[Reference]

[1] EBU,"Specifications of the Radio Data System (RDS) for VHF/FM Sound Broadcasting",Tech.3244-E,1984.

[2] CCIR Document WP10/5(1991),"A New Modulation Method of FM Multiplex Broadcasting for Mobile Receivers",1991.

[3] O. Yamada,"Development of an Error-correction Method for Data Packet Multiplexed with TV Signals",IEEE Trans. on Communications, Vol.COM-35, No.1, pp.1045-1055, 1987.

[4] S. Moriyama,etc.,"Traffic Information Services using FM Multiplex Broadcasting",IEEE, VNIS'92, Session 16, pp.434-439, 1992.

Table 1 Transmission parameters (final values)

Modulation method	L–MSK
Subcarrier frequency	76kHz
Bit rate	16kbps
Multiplex level	4–10%*

*

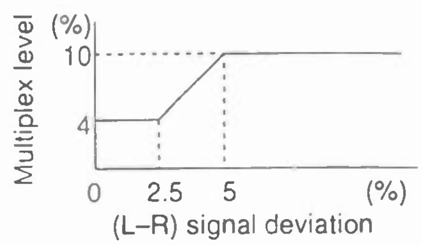

Table 2 Service model

Service	Kinds of information	Total data quantity	Transmission capacity*
Level 1	Text display (15chara.x2row) Traffic jam in local area Highway information Traveling time Traffic regulation	18kbyte	1kbps
Level 2	Graphic display (248x196dots) Level 1 text information Map of traffic jam	38kbyte	2kbps
Level 3	Graphic display CD–ROM maps Level 2 information Detail traveling time data	99kbyte	5kbps

* : Parity check bits for error correction not included. To include parity check bits, the total bit rate is 2.3 times that of transmission capacity.

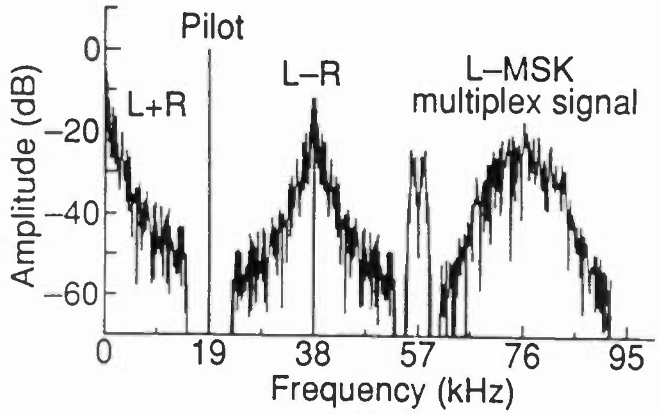

Fig.1 Baseband spectrum of FM multiplex broadcasting

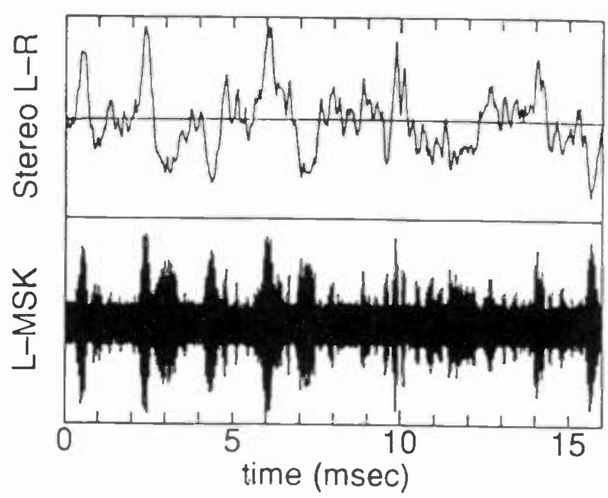

Fig.2 A waveform of L-MSK signals

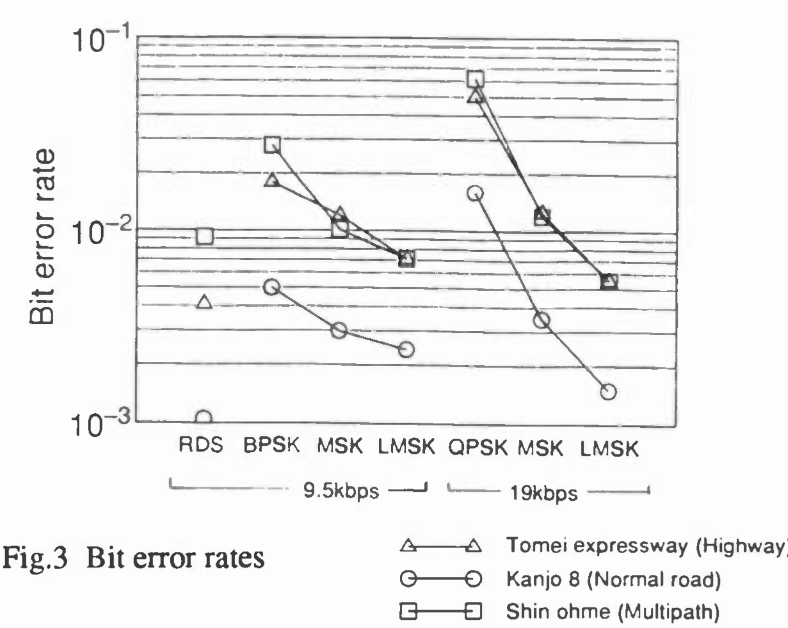

Fig.3 Bit error rates

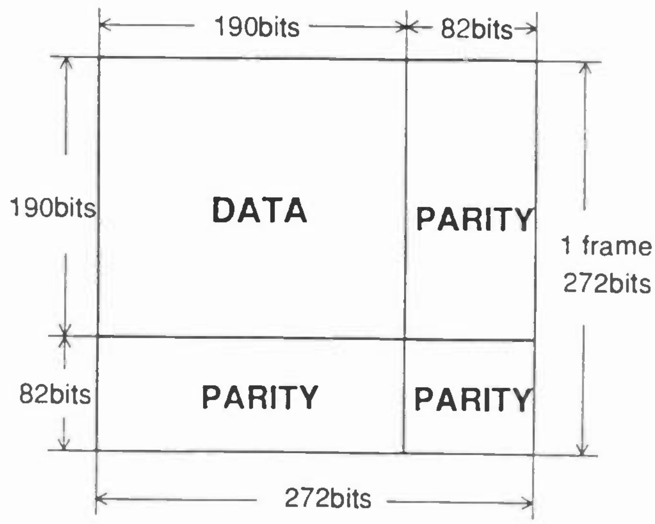

Fig.4 Frame structure

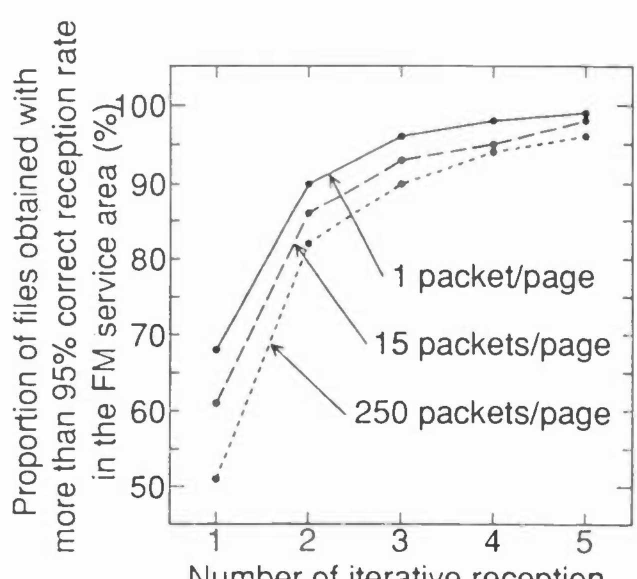

Fig.5 Iterative transmission effects

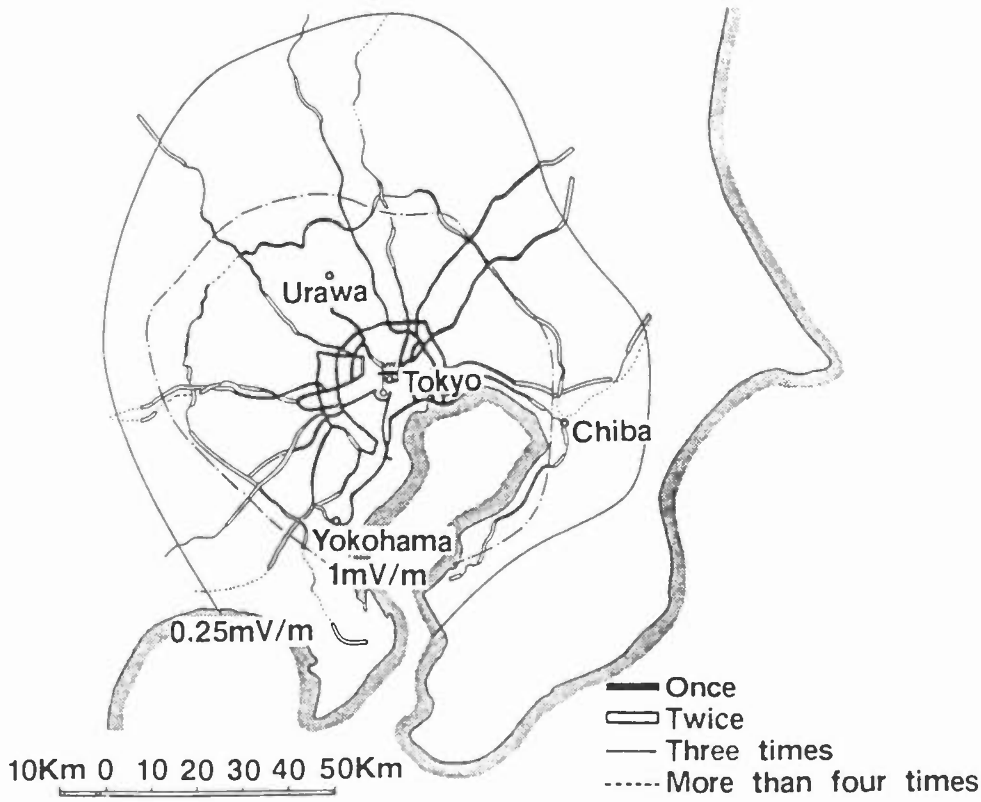

Fig.6 Routes having more than 95% correct reception rate

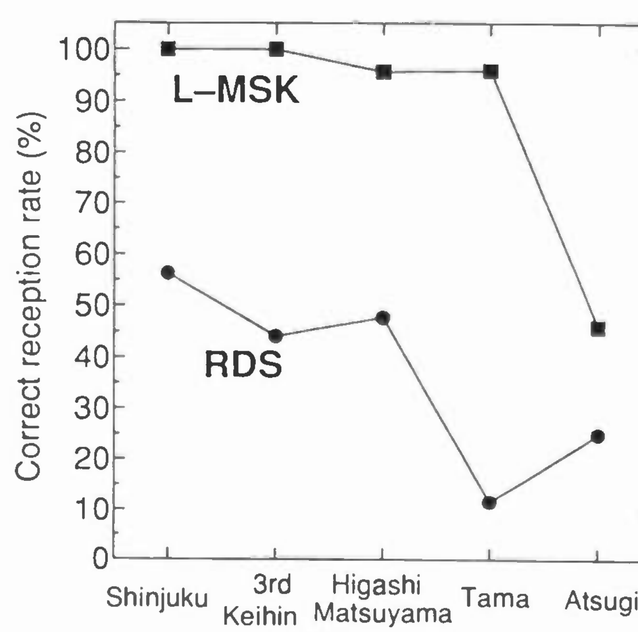

Fig.7 Comparison between L-MSK and RDS

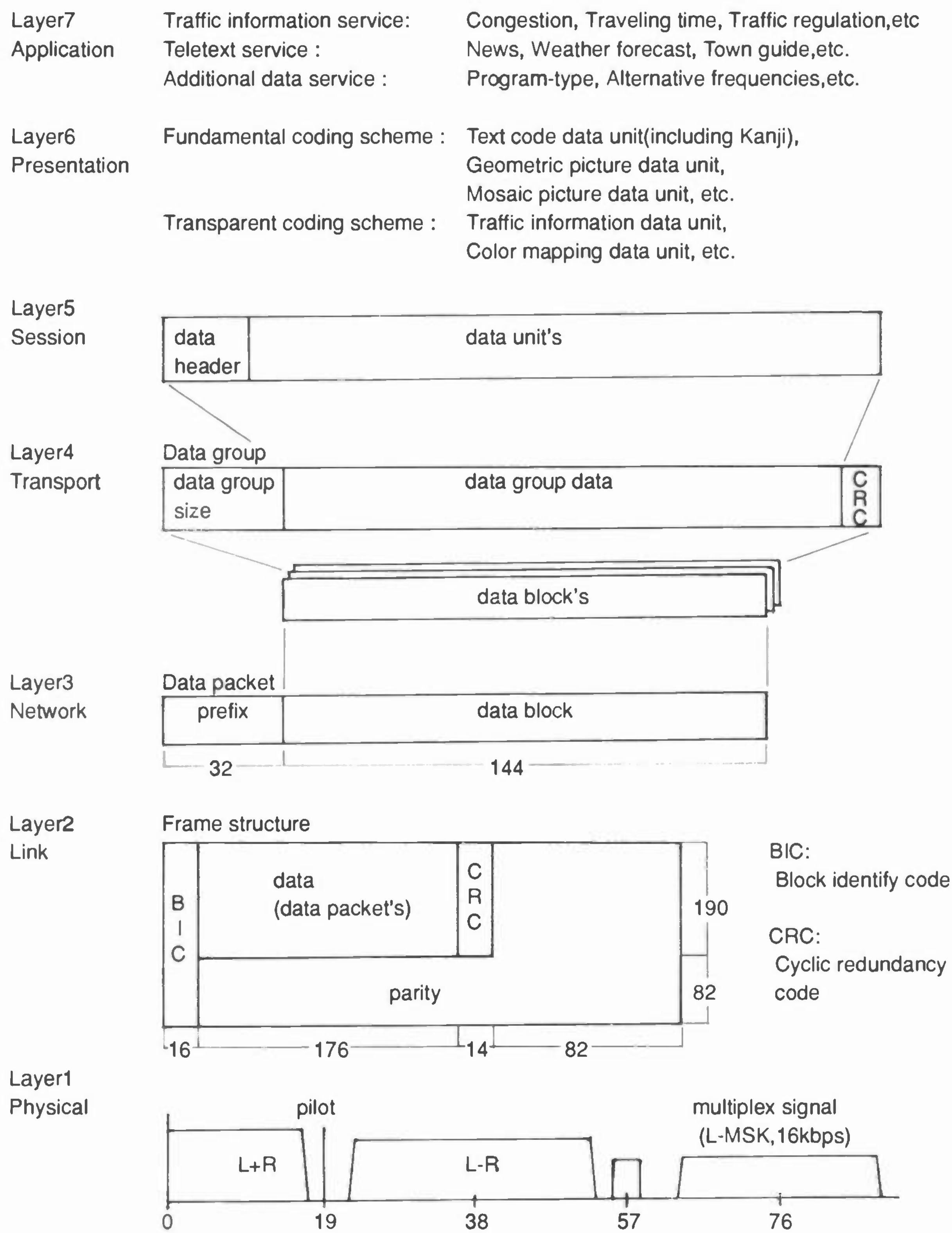

Fig.8 Outline for system protocol

DATA BROADCASTING: TV

Wednesday, April 21, 1993

Moderator:
Bob Ogren, LIN Television Corporation, Providence, Rhode Island

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*Paper not available at the time of publication.

VBI DATA BROADCASTING: NEW REVENUES FROM AN UNDER-UTILIZED ASSET

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Rapidly evolving technology and the increasingly competitive business environment are raising awareness within the broadcast community of expanded business options for its established information distribution infrastructure. Many proactive broadcast organizations are focusing their attention on low-cost high performance VBI data broadcasting technology to gain a foothold in this new business field by reselling surplus bandwidth and/or implementing value-added services

The purpose of this paper is to provide an overview of VBI data broadcasting basics, a summary of the key applications of point-to-multipoint data broadcast delivery, and guidelines for implementing data broadcasting.

OVERVIEW

VBI data broadcasting is a proven, high speed, point-to-multipoint data transmission medium. Data is carried with a standard television signal in unused vertical blanking interval (VBI) bandwidth and can be received everywhere the signal reaches. For data transmission applications requiring delivery to many destination sites, data broadcasting offers compelling cost savings, unmatched timeliness of delivery, and comprehensive geographic coverage.

The vertical blanking interval is the black stripe seen when a television picture loses vertical hold and rolls. It is an integral part of the NTSC television standard and makes up the first 21 lines of each field The VBI is black because it is empty; it's part of the signal but carries no information. Because this bandwidth is not used, it is available for data transmission. For example, line 21 of the VBI is widely used to deliver closed captioning information.

VBI data broadcasting (once known as teletext) is a technique developed in the early 1980's to utilize lines 10 to 20 of the VBI to transport virtually any type of digital data along with the television signal. With the recent development of low cost VBI hardware and the explosive growth of electronic information, data broadcasting technology has finally come of age.

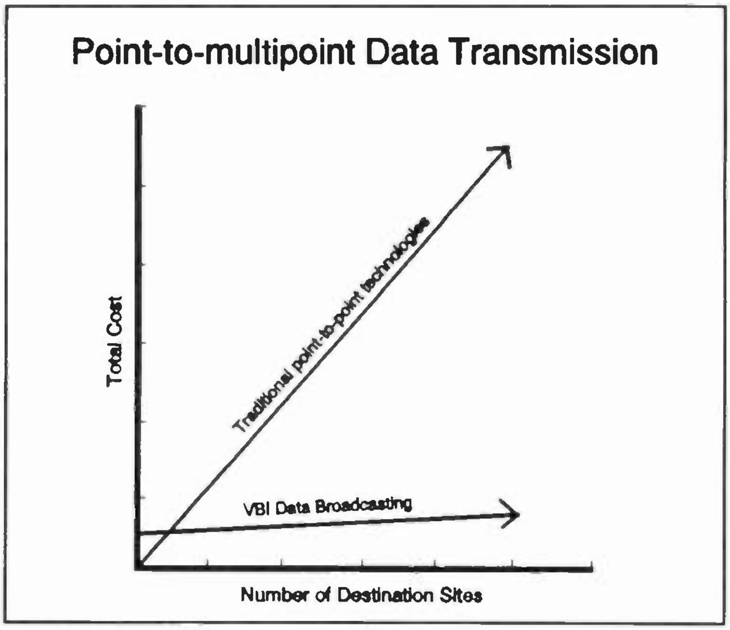
Because the data is actually encoded as video, it becomes an integral part of the television signal. The VBI data is available everywhere the television signal reaches and will pass transparently through all carrier media. VBI data transmission technology is well established in Europe and is on a fast growth track as a niche data transmission alternative in North America.

The industry technical standard for VBI data broadcasting/teletext is NABTS (North American Basic Teletext Specification) an EIA/ANSI standard (EIA-516). NABTS specifies the essential technical requirements necessary to ensure that transmission and reception equipment from compliant suppliers will work together.

The television VBI has been deregulated by both the FCC in the U.S. and the CRTC in Canada. Major national television networks, as well as numerous regional and niche broadcasters in both countries are currently exploiting this technology to deliver data along with their television signals.

CAPACITY & DATA INTEGRITY

VBI data broadcasting is a high speed in-place transmission medium with a data input capacity of up to 185,000 baud. Data can be delivered simultaneously to an unlimited number of reception sites located wherever the signal travels. Individual data inputs, each up to 19,200 baud asynchronous, are carried in a packet multiplexed format to utilize the available bandwidth. Because the system transmits data in only one direction, it employs sophisticated forward error correction algorithms to guarantee data integrity.



Compelling cost savings versus traditional point-to-point technologies

Each line of the VBI is capable of carrying 288 bits per television field (one packet). Of this, 64 bits are used for set-up and addressing, leaving 224 bits for data. At 60 fields per second and 11 VBI lines per field, this translates to a data capacity of 147,840 bits per second (bps). Because data is input over an RS-232 asynchronous link, but the start/stop bits are not carried over the VBI, the maximum input capacity is about 184,800 baud (147,840/0.8) for 11 VBI lines. Fewer VBI lines will yield proportionately lower capacity.

State-of-the-art data broadcast "transmitter" delivery systems provide maximum flexibility to subdivide this bandwidth resource amongst a large number of simultaneous users. Data from multiple sources are transmitted concurrently employing sophisticated packet multiplexing techniques; each data stream travels on its own virtual private circuit.

Advanced addressing structures provide the flexibility to deliver any single data file to all user group sites or to only a specified group, sub-group, etc.. With this power, the scope of potential end-user applications is limited only by the imagination. In addition, encryption techniques are easily implemented for highly sensitive data.

In addition to standard VBI transmission, some data insertion systems are capable of inserting data into all or any combination of the entire 525 line video channel for private networks or after hours use. This "full field" transmission enables over 4 megabits per second throughput capacity.

COST

A major attraction of the VBI transmission medium is that it is free (or at least already paid for) because the data is carried over existing infrastructure. No additionaltransmission equipment or power is required, even as the number of receiver sites increases. This point-to-multipoint leverage typically enables very significant cost savings compared to alternative technologies. Utilization of this in-place VBI asset requires a data insertion system at the television network "head-end" and an inexpensive VBI receiver at each site.

Each incremental reception site requires only a receiver unit and a source of the television signal. Sophisticated receivers (essentially one-way modems) range from data-only models to data/graphics versions capable of receiving and then displaying graphics/text overlays. The television signal is typically accessed through a conventional television antenna, a cable system, or a satellite dish.

For applications "natural" to VBI data broadcasting, typically characterized by many reception sites, sensitivity to the timeliness of information delivery, and disperse geographic coverage, the total cost is very low relative to alternatives (if any in fact exist).

KEY CHARACTERISTICS

VBI data broadcasting offers a very cost-effective and timely means of transmitting electronic information from a central point to multiple destination sites. Content is broadly defined and includes text, graphics, software, or virtually any other information in electronic form. The data can be accessed anywhere and any way the television signal is available. Six key performance characteristics serve to define the most suitable applications.

Point-to-multipoint

One of the key advantages of data broadcasting is that, with a single transmission, the data is available everywhere the television signal travels . . . literally everywhere. With the telephone system, transmission costs increase roughly in proportion to the number of destination sites. Using data broadcasting, however, the incremental cost of delivering the same data to each additional reception point is zero; one transmission can reach a potentially unlimited number of users. This inherent leverage offers very significant cost savings when data requires delivery to multiple reception sites.

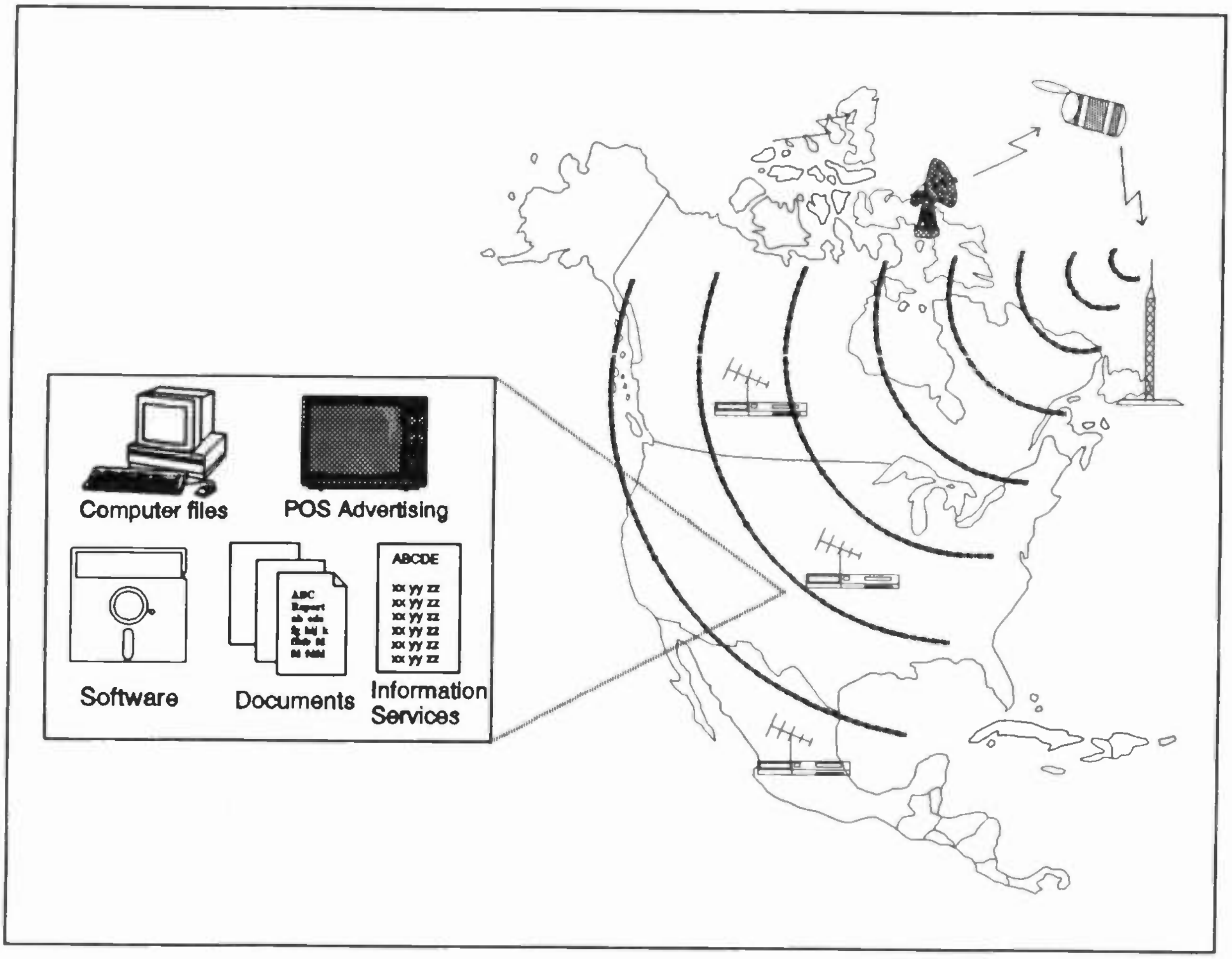
Timeliness

The second fundamental characteristic of data broadcasting is the timeliness of data delivery. For many sites, data broadcasting is the fastest communications medium available because a single transmission reaches all sites. In addition, data broadcasting provides simultaneous delivery to any number of receivers. Telephone systems by design can only transmit to a single site at one time.

For example, a 5 megabyte file requires about an hour to transmit over an asynchronous 19.2 Kbps "telephone" link. Transmission of the same file to hundreds of sites would require many days and/or duplication of expensive hardware and transmission lines. With data broadcasting, however, the same file can be delivered to each and every site, regardless of the number, in just one hour.

Broad geographic coverage

Television is often referred to as the most pervasive communications technology available. It delivers "the last mile" link virtually anywhere and everywhere. Although the telephone system is an extensive national network, its reach and capacity is limited to the presence of physical plant. Data transmission over telephone links is very expensive or often not available



Comprehensive geographic coverage and timely delivery for point-to-multipoint data transmission for many non-urban sites. The incremental cost of adding "the last mile" to the telephone network is astronomical relative to the inexpensive television antenna required to access a VBI data broadcast.

Addressing Security and Flexibility
Although the data travels everywhere along with the signal, only receivers preprogrammed with the appropriate address authorization code(s) can access a specific transmission. The multiple levels of addressing available, similar to techniques developed for electronic mail systems, also optimise applications flexibility. Information delivery can be dynamically addressed either to all sites in the user-group, to one or more sub-groups, or even to a single site.

Output media
At each site, the VBI receiver can deliver addressed data directly to many types of devices. Typical examples are:

- database, files, or software delivery to computer,
- text/graphics overlay at display units or cable heads, and
- document hardcopy on a printer.

Reliability
Broadcast networks are typically among the most rugged, redundant, and reliable systems in the world, typically delivering at or near 100% up-time. In addition, broadcast networks, with limited points of failure, are more immune to natural disasters (such as earthquakes) than are terrestrial systems.

The data integrity of VBI data broadcasting is as good or better than traditional telephone system data lines. With advanced forward error correction, delivery performance is significantly better than published transmission specifications for point-to-point telephone technologies.

One-way Transmission
The primary limiting factor in data broadcasting is that it is a one-way transmission system. There are, however, many compelling niche applications suited to one-way delivery. In addition, applications which have the majority of data transmitted in one direction with low volume return communication, can use a hybrid system; VBI data broadcasting carries the bulk of data one-way, while a modem link carries the limited return.

APPLICATIONS

Corporate & Government Data Transmission
VBI data broadcasting technology is a natural match for many corporate government applications. These organizations rely heavily on telecom-based data transmission to move massive quantities of "data" around a broad geographic area. Four applications examples are described below according to the type of data being transmitted.

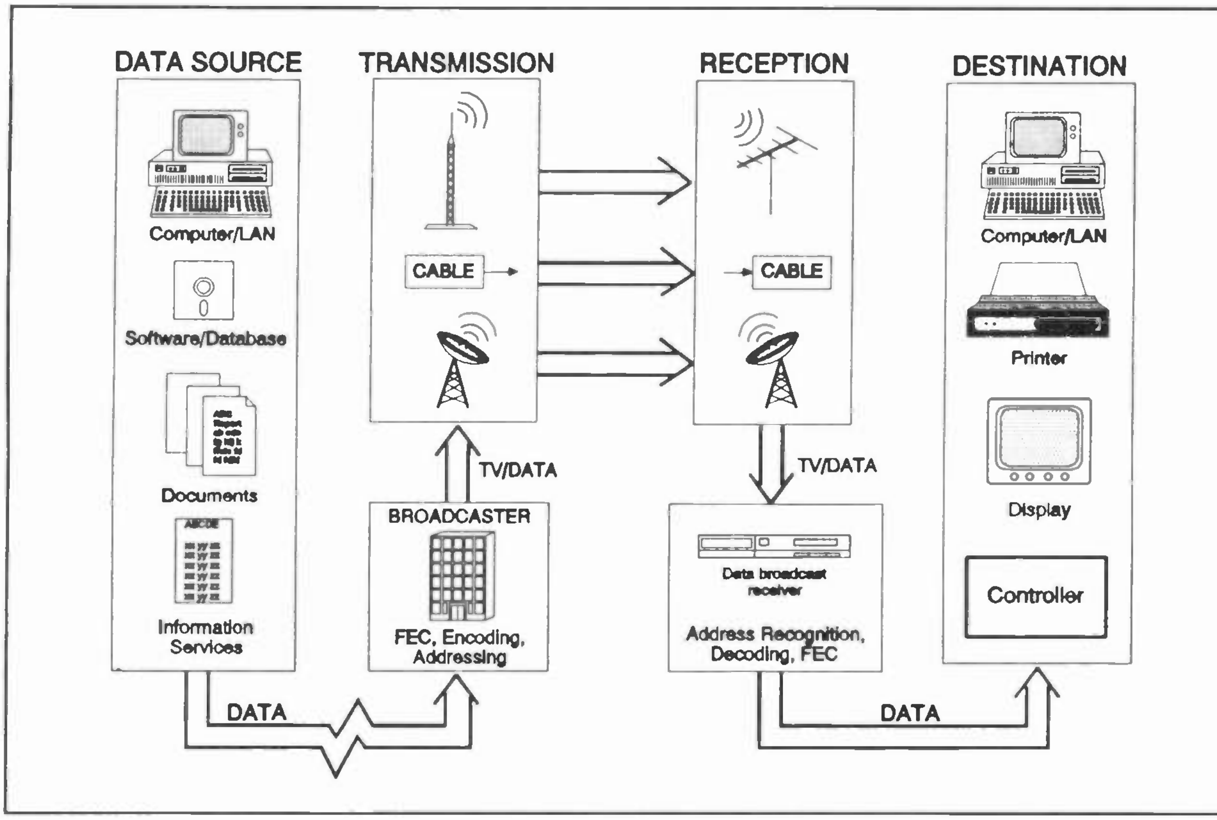
Computer software: An MIS executive with many users of specific software (any company with a large branch network) has few options for distributing updates. Although transmission over data telephone lines or via physically transported floppy disks are expensive and cumbersome processes, the major problem is one of control. Particularly for large software files, it can easily take many weeks to update everyone in the system. In the meantime, all sites are running on different versions, an undesirable situation.

Data broadcasting offers an attractive solution to this problem. Not only does it offer large cost savings, but each site receives the updated software simultaneously, and in minutes/hours and not days/weeks.

Documents: Currently an almost unimaginable array of printed material is distributed on a regular basis at considerable shipping expense and with often unpredictable timing. Data broadcasting offers the capability to distribute any document in electronic form to any number of sites simultaneously, anywhere in the country, and faster than any other communications medium. Documents can be delivered automatically to computer memory or direct to a printer.

In addition, organizations increasingly prefer to manage documents in electronic form, but distribution to multiple sites by telecom is expensive. Some major organizations physically circulate computer generated and printed documents to end users who, in turn, actually use electronic scanners to reformat the information into electronic form. VBI data broadcasting technology provides the missing link to enable the most economical and effective distribution of documents from a central source to many users.

Distributed database: When faced with the dilemma of providing access to a central database to many potential users, conventional thinking implements a costly central computer and multiple dial-in telephone lines. When many users are distributed across a broad geographic area, this results in sizable time-based



Data Broadcasting Transmission Path

access charges and slow system response under peak demand. As usage increases, additional capital must be invested to increase capacity.

Although data broadcasting is not the solution to all applications, there are many where a shift in thinking to a distributed database approach can offer improved service at lower cost. Instead of a single central dial-in database, the data can be simultaneously broadcast via the VBI to computer memory at each site. Here the end user can access and use the information at his/her leisure, and at no incremental cost.

Other data: In addition to the specific applications cited, virtually any type of data/information is a candidate for VBI data broadcasting if it does or could exist in electronic form and is distributed primarily from one point to multiple destinations. In some "two-way" transmission applications, data broadcasting is used to transmit the bulk of information flow out to multiple destinations, while dial-up modems handle the limited return, resulting in significant cost savings versus a telecom-only approach.

Major organizations with multiple sites spread across a broad geographic region are the most logical candidates for VBI data broadcasting. Financial institutions (banks, trust, insurance, and securities companies) and retail organizations are the largest identifiable groups. Data broadcasting is almost by definition an ideal match for governments organizations. They have a large quantity of information generated/stored centrally and require distribution across a broad jurisdictional area. In addition, the ability of VBI data broadcasting to deliver quickly and simultaneously to all recipients, can be a major contributor to guaranteeing equitable opportunity to all.

Information Services

Information services which include financial information and other electronic publishing, represent a distinct applications segment for data broadcasting technology. Although the technology represents a cost effective means of distributing the information when many destination sites are involved, the primary perceived benefit in this segment is often the timeliness of delivery.

This "timeliness" refers to both the unmatched speed of delivery to all sites, and the simultaneous reception by all sites. No other transmission medium available can offer this level of performance.

Advertising & Public Information

VBI data broadcasting technology is currently being used to download data and graphical information directly to networks of monitors, LED signs, and other display devices. As examples, content has included information such as financial rates in bank branches, advertising-based news/weather/sports/stock quote services, advertising and product promotions in retail stores, and travel schedules in airports, hotels, and convention centres.

The ability to deliver text/graphics information direct to an inexpensive television or RGB monitor without the need for a computer, is a powerful feature of the most sophisticated VBI receivers.

Interactive Television/Multimedia

As most broadcast professionals are aware, an ever-increasing array of "interactive television/multimedia" services are being conceived and implemented for the home television viewer. Typically categorized as one of television programming schedules, home entertainment, or transaction services, these services usually combine a number of computing, communications, and presentation technologies to achieve the desired capabilities and performance.

Because of the unique capabilities of VBI data communications, namely its point-to-multipoint delivery capability and obvious timing and physical synergies with television-based applications, many of these services have integrated VBI communications as an information delivery mechanism.

Education

Because of its low cost for multiple sites and pervasive geographic coverage, data broadcasting is an attractive transmission alternative for many educational organizations.

K-12 school districts use data broadcasting technology to distribute both administrative data (such as electronic mail, teaching methodologies & manuals, restructuring information, etc.) to office systems, and databases and software updates direct to classroom computers. Because of the strong synergies in transmission medium and point-to-multipoint communications, VBI data broadcasting is also being integrated with distance education programs to deliver course materials such as databases, software tools, documents, etc. along with the educational transmission.

Power Load-Shedding

One of the key dilemmas of any electrical generating utility is that capacity must be in place to handle peak period demand. Because peak demand is significantly higher than base or average demand, a utility must install and maintain capacity that is not generating revenue most of the time.

For years utility executives have dreamed of the ability to reduce peak demand by turning off devices which do not need to operate during peak periods. The utility is capable of tracking and predicting the power usage and peak periods. It can identify non-essential users such as industry with dual power sources, industrial water heaters, etc.. Furthermore, these users are receptive to incentives to allow external control of these devices. The missing link has always been a reliable and timely means of communication with a large number of geographically disperse sites.

VBI data broadcasting has filled this communications gap. It provides a highly reliable transmission medium, capable of delivering instructions to power controllers at sites located virtually anywhere.

SUMMARY

The television VBI is an in-place, deregulated asset which is often underutilized by broadcast organizations. It's use is well established overseas and is rapidly being implemented across North America. National, regional, and specialty television organizations have established transmission capability and are reaping the benefits of this unused asset for a number of profitable applications.

Data broadcasting over the VBI is a niche, but powerful data transmission alternative. For applications requiring delivery of information from one point to many sites, it offers compelling benefits highlighted by:

- substantial cost savings,
- unmatched timeliness of delivery, and
- comprehensive geographic coverage.

The technology to capitalize on this available bandwidth is proven and available now. Increased market demand and an expanding array of applications have driven implementation costs down. If you are a broadcaster with an underutilized VBI asset, you should be aware of the compelling capabilities of VBI data broadcasting and the potential for new revenues and profits.

EMERGING BUSINESS OPPORTUNITIES THROUGH CAPTIONING IN THE NINETIES

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INTRODUCTION

The world, as always, is evolving to a new age. The future promises great hope for the many who fall outside of our typical societal norms. Today, a light shines through where technology can bear fruit and opportunity will open a new world to many more people. Closed captioning is one of those technologies. In July of 1990 congress passed into law, (yes, those politicians), some of the most sweeping human rights legislation in history, the Americans with Disabilities Act and the Decoder Circuitry Act.

What does this mean? Well, basically, it means we finally have laws that protect the rights of a very large and sometimes forgotten minority group– people with disabilities. This represents some forty three million Americans. Those are big numbers. The largest segment of this group is the deaf and hard of hearing, representing twenty four million people.

THE LAW

Title II of the ADA states:

"(1) A public entity shall furnish appropriate auxiliary aids and services where necessary to afford an individual with a disability an equal opportunity to participate in, and enjoy the benefits of, a service, program, or activity conducted by a public entity.

(2) In determining what type of auxiliary aid and service is necessary, a public entity shall give primary consideration to the requests of the individual with disabilities."

28 C.F.R. Part 35, 56 Fed. Reg. 35694 (July 26, 1991)

According to the Justice Department regulations (the governmental body charged with enforcement of ADA Title II), auxiliary aids and services are defined as: *"...closed caption decoders, open and closed captioning, videotext displays, or other effective methods of making aurally delivered materials available to individuals with hearing impairments."*

Public entities are also defined under Title II as: *"...any State or local government; any department, agency, special purpose district, or other instrumentality of a State or States or local government."*

Other titles of the ADA have provisions concerning private entities. In fact, it is estimated that over five million private establishments will be affected.

The Decoder Circuitry Act, also passed in 1990, will require built-in decoder circuitry for every television set manufactured or sold in the United States with a screen 13" or larger. The implications of this act are sweeping. It will benefit not only the deaf and hard of hearing population, but will also benefit the twenty seven million Americans learning English as a second language, twenty seven million functionally illiterate, and thirty million elementary school students who are learning to read.

THE MARKET AND OPPORTUNITIES

Everything from national programming, movies, local news, sporting events, church sermons, government meetings, corporate trainings, and much more is being captioned today. The market is exploding and opportunities are abundant for those who wish to participate. But, these devices are not limited to only serving the world of captioning. Now underway is specification for a whole new channel of information in the second field of VBI (vertical blanking interval) line 21, the implications of this are great, and we will soon see new products and services emerging- everything from electronic TV guides to home news and stock services.

How do you get started?

Well, it depends on the type of captioning and how it will be displayed. Open and closed captioning are the two primary methods in use today for display. Open captioning is the least expensive way to caption, requiring less expensive equipment, and the captions are permanently written on the video frame. Open captioning is oftentimes the preferred method for people captioning meetings and smaller church services. Closed captioning, on the other hand, requires the use of both an encoder and a decoder, and the information is coded on line 21 of the vertical blanking interval.

For those of you who are not familiar with the term VBI (vertical blanking interval), the vertical blanking lines consists of those black lines you see on older television sets when the video signal is rolling and out of adjustment. Most newer sets have special circuitry to correct this problem.

CAPTIONING SYSTEMS

The two types of captioning systems presently sold commercially are online and offline systems.

Online Captioning Systems

Online systems are most often used for live on-air environments such as news, sports, meetings, etc. A full system consists of a live television feed, a television monitor, personal computer, software, steno machine (much like those used in modern day courtroom environments), an encoder and decoder in Figure 1.

Online Captioning System Software The heart of the online system is the software that controls it. Online systems are based on a stenographic-type translation system. A steno machine, similar to those used in courtrooms, is used to input the data- which in this case is the "shorthand-like equivalent" or steno of the spoken word. The Stenocaptioner produces this information by stroking it on the steno machine. The steno is then translated by the realtime translation engine into its English equivalent. Intelligent systems have special capabilities such as a phoneticizer that will translate undefined dictionary entries or strokes into their phonetical equivalents.

All the companies that presently manufacturer such systems emerged from the world of CAT, computer-aided-transcription, in courtroom environments. These CAT companies were best suited to develop this market because the translation engines were already developed for their courtroom applications. Steno input is required because it is the only method available today for accurately keeping up with the spoken word. Online captioning requires a Stenocaptioner to write at the speed of 200 - 250 words per minute, which is faster than even the fastest of typists.

Top of the line systems also have a number or other features such as the ability to make new dictionary entries quickly and changes on the fly. The ability to make changes quickly is critical in live applications such as newsroom environments where these systems were first applied. A good online system will also have the ability to switch between pre-scripted information (or files) and steno feeds. In newsroom environments, the ability to pre-script information is particularly important where all information, with the exception of live segments, is usually written and produced beforehand, relieving the Stenocaptioner from having to stroke the information for the entire newscast.

In news environments, the captioner is oftentimes assisted by a coordinator who assists the captioner by performing activities such as changes in feeds, new dictionary entries, global-word-change entries, story queuing, formatting changes, queuing of dictionaries, and other maintainence activities. A Stenocaptioner goes through the same training program as a Court Reporter requiring a number of years of formal study and practice

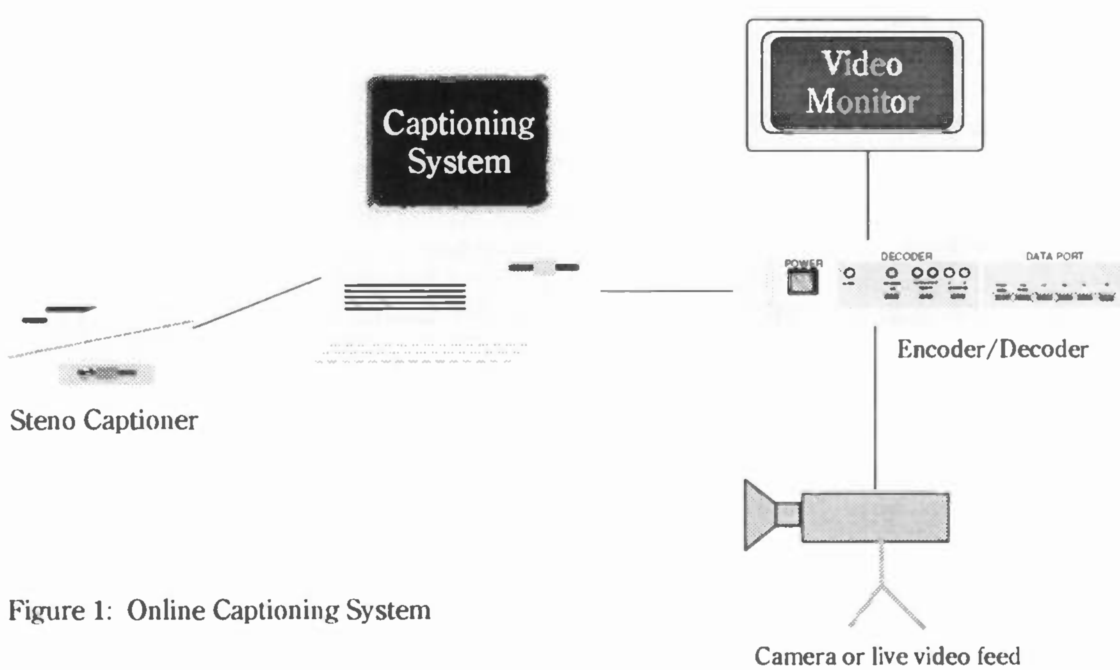

Figure 1: Online Captioning System

(normally 2-5 years) followed by a rigorous certification exam. The drop-out rate is very high, normally only one out of ten people make it through the program. Most court reporting schools, however, do not prepare them for the demands and high accuracy levels required for realtime environments. A number of months (usually 3-6) of additional training and practice is required to reach accuracy levels required for live broadcasts.

Other very necessary features for online software include: 1) The ability to support stock captions for information like credits that are displayed at the end of the program. 2) Programmable format keys that allow you to change the caption display formats quickly. 3) A history window that allows you to view captions that already have been sent and thereby make global changes quickly when a word is not stroked correctly. 4) The ability to support multiple customized dictionaries. 5) The ability to support global word changes. 6) The ability to quickly search the text of pre-scripted stories and make changes on the fly. 7) The support of all existing hardware encoders and decoders. 8) The ability to save information in portable ASCII formats. 9) Support of the numerous types of steno machines. 10) The ability to support all the various caption types, special characters, special effects, and formats. 11) The ability to block or pass captions, and the ability to support pre-sending modes for pop-on captions. 12) The ability to queue a number of stories in succession and feed them or change the order as the news director demands. Online captioning software is quite sophisticated and very few vendors offer products that can support the demands of live environments.

Online Captioning System Hardware On the hardware end, the story is somewhat simpler. Most modern day steno machines, used by trained personnel, Stenocaptioners and Court Reporters, are adequate if equipped with the ability to support serial data outputs. A serial cable is then used to feed the realtime translation software running on a personal computer. Once the steno is translated, it is fed to a hardware encoder device that writes the appropriate codes in line 21 of the vertical blanking interval of the video signal. A decoder is then used to interpret the code on line 21 and display the captions. Older decoder devices support the old display standards that divide the video display into a 15 row by 32 column character matrix, allowing characters to be displayed only on the top or bottom four rows

The newer decoding devices allow for a much wider feature set. The new decoders also support: 1) full screen support for captions; up to four rows at a time;
2) an extended character set; and
3) paint on captions. Proposals are currently under discussion for reserving a second field, line 21 field 2, and expanding its capabilities to support not only captioning and text, but also extended data services.

Offline Captioning Systems

Offline systems are most often used for off air environments where more time is available for perfecting results. Offline systems are used for captioning movies, documentaries, and any other type of pre-taped broadcasts. A full system consists of a one or two VTR devices, a full motion video card, a time code card, one or two television monitors, a personal computer, offline captioning software and an encoder and decoder as pictured in Figure 2. A steno machine and CAT software are optional.

Offline Captioning System Software Much like online systems, most of the user interface and control options are available through the software which integrates and controls the entire system. The offline software system is set up to take input from a keyboard or a pre-scripted ASCII file. The best method for inputting data is again having a Stenocaptioner or a Court Reporter stroke in and edit the audio using an online system or any standard CAT system that can output ASCII file formats. The next step would be to import the data and to use the offline system for placement, customization, timing and display.
A high end software system has a number of characteristics that differentiate it. The key here is productivity. The job of doing offline captioning is quite labor intensive. Up until recently, captioning a one hour tape could take twenty to forty hours depending on complexity of the project. Most of the captioning was being done through three main captioning service providers who had enough volume to support in house software development. Over the past year, several vendors have announced or delivered products that have not only opened up the market to more companies, but some of these products promise to reduce the number of hours required for delivering a offline captioned tape by a factor of four or five.

Tape captioning times can vary dramatically depending on the sophistication required for the project. For example, some tapes might employ simple four line roll-up captioning without any special placement of captions on the

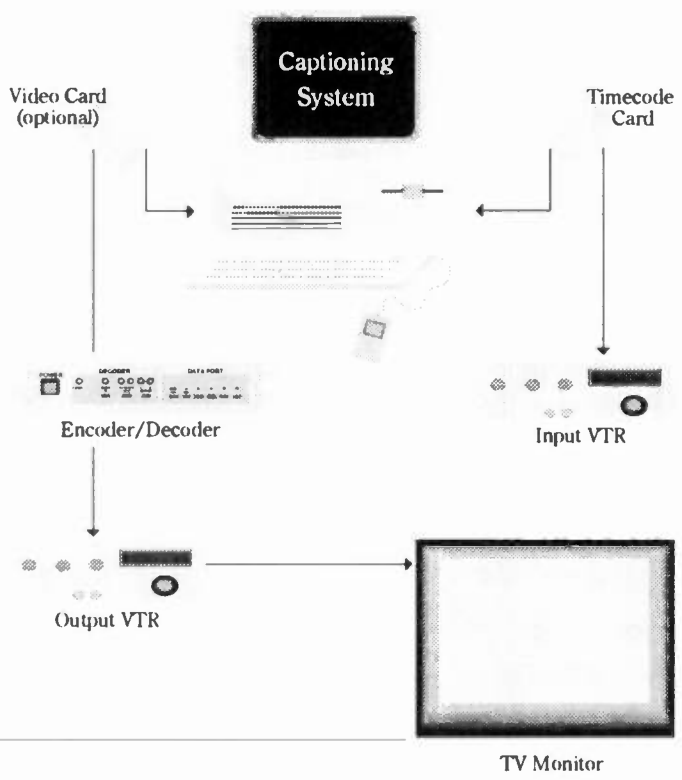

Figure2: Offline Captioning System

tape. Someone might do this on a training tape to reduce costs. On a project where budget is less of an issue, such as a high-end movie, more time intensive captioning might be required using sophisticated techniques such as intermixed pop-on and roll-up captions, multiple on-screen caption placement for different speakers, use of special character attributes and other effects which drive up the costs.

In order to dramatically reduce the time involved for offline captioning, a number of features must be present in the software. These include: 1) Software-remote VTR controls, freeing the caption producer from the laborious task of manually operating a VTR; 2) An easy-to-use windowing interface with pull down menus; 3) Context sensitive help; 4) Caption positioning through mouse control; 5) Block cut and paste functions with timecode re-sequencing capabilities; 6) On-screen caption preview and simulation; 7) Rapid text search; 8) Sophisticated style and attribute control; 9) Full support for special character sets; 10) Support for multiple encoder/decoders and VTR and tape formats; 11) Video in a window on the computer screen.

Offline Captioning System Hardware The story around Offline Captioning System hardware is again less complex featurewise than the software. The VTR devices should be of industrial grade, should be SMPTE timecode based and equipped with either RS232 or RS422 serial controls. A built-in or external time-based corrector is recommended for complete editing environments where the deck will serve as a source. A full system will require two VTRs (source and record). A single lower end deck that is frame accurate and has computer controls (without a built in TBC) can be used if the actual production work or encoding is to be done offsite. The personal computer to be used should be a higher-end system (such as 386DX or 486DX). The computer or the tape deck should be equipped with a timecode reader. A PC video card is recommended for on-screen preview and fast editing. The encoder and decoder are identical to those used for Online Captioning System. At least one external monitor is recommended to view the result.

SUMMARY

Many opportunities will be appearing in 1993 for those interested in captioning. In July of 1993, both the Decoder Circuitry and the portions of the Americans with Disabilities Act primarily concerned with captioning will take effect. For those interested in providing captioning services, two types of systems exist - online and offline captioning systems. Online systems are used in live display environments where making changes on the fly is most important. Offline systems are used in more labor intensive and less time critical projects where the most important element is productivity in order to reduce labor costs. A high-end Online Captioning System normally costs around fifteen to twenty thousand dollars. A high-end Offline Captioning System also costs normally around fifteen to twenty thousand dollars plus the cost of the VTRs.

DATA BROADCAST: NEW OPPORTUNITIES IN THE VBI

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This paper is a survey of the Extended Data Service capabilities created by the EIA-608[1] standards process for line-21 data systems. Recent FCC actions concerning the authorization of the field-two signal as an additional data signal are summarized. A number of new service opportunities for program producers, television networks and local broadcasters are described. Basic signal handling requirements and encoding equipment needs will be presented in a companion paper[2].

INTRODUCTION

EIA Involvement:
The EIA became involved in the closed captioning standards processes with the passage of the TV Decoder Circuitry Act of 1990. After a joint industry task force completed its recommendations to the FCC on TV caption decoder display standards, this task force became known as the Television Data Systems Subcommittee (TDSS) and began work on the "fun-stuff." This process included the development of an industry standard for interleaving Extended Data Services (EDS) into a captioning signal as well as new and enhanced captioning modes. The new services were developed with the intention of using them exclusively in the previously unused portion of the line. This provides for the additional bandwidth and prevents interference with existing caption decoders.

Table 1 lists the sequence of events in the timeline for the EIA-608 standards process.

FCC Involvement:
On the basis of FCC rules for line-21 (in place at the time this was written), there is the authorization to carry non-program related data[3] within the line-21 waveform. The TEXT services have been an example of this authorization. This authorization would probably allow other new data services within the line-21 waveform including EDS. The catch is, that they must conform to the established data format. Even though it is permissible to do so, it is not practical, due to the limited amount of bandwidth available in the existing data stream.

The FCC rules did not, however, allow full use of the field-2 waveform. Only one-half of the line-21 on field-2 was defined[4]. This limitation restricted the possibility of ever duplicating the full line-21 waveform on both fields. Current line-21 encoding equipment generates these half-line test signals, but they have never been used in any decoding hardware. In reality this entire line has gone to waste.

In July of 1992, the EIA petitioned the FCC to allow full use of line-21 on field-2 for captioning formatted data signals. By authorizing this additional half-line, the FCC will allow a host of new TV data services without any problems of interference to existing services.

MILESTONES of EIA Line-21 EDS

1990	EIA/NCTA Joint Engineering Committee considers line-21 data format for program and source identification.
September 1990	EIA Caption Display Standards Joint Task Force is created.
October 1990	Congress passes Television Decoder Circuitry Act of 1990.
December 1990	EIA Task Force submits proposed receiver display standards for captioning to the FCC.
January 1991	FCC issued Notice of Proposed Rulemaking in response to Decoder Act.
April 1991	FCC issues Closed Caption Decoder Requirements for Television Receivers (91-119).
September 1991	The goals and objectives of the EDS system are outlined by TDSS.
October, 1991	Strawman specification of Line-21 EDS system is submitted to TDSS.
April 1992	FCC issues Memorandum Opinion and Order further clarifying captioning requirements (92-157).
July 1992	EIA petition to FCC for authorization of line-21, field-2 for data services.
August 1992	Mitsubishi produces first Laser disc with line-21 EDS.
August 1992	WGBH & MARDO produces first test tape with line-21 EDS packets.
September 1992	First interim draft of EIA-608 published.
October 1992	EIA introduces new EIA-608 features to industry in Workshop.
December 1992	FCC issues Notice of Proposed Rule Making Process on VBI revisions.
-----------------------	**FORECASTED EVENTS** -----------------------
April 1993	First over-the-air transmissions of line-21 EDS material.
April 1993	First public demonstration of TV receiver with EDS capability.
June 1993	TDSS releases EIA-608 to parent committee for comment.
September 1993	Final amended EIA-608 is released for full membership vote.
November 1993	If approved, EIA-608 is recommended to ANSI as formal standard.
1994	Widespread market introduction of products with EDS capabilities.

Table-1. History of Line-21 EDS

This request for changes to the line-21 VBI signal was combined with a simultaneous request for the use of line 19 as the ghost cancellation reference signal when the FCC issued the Notice of Proposed Rule Making on December 31, 1992[5].

It should be emphasized here that if the FCC acts favorably to the EIA request, any use of the field-2 signal is voluntary. There will be no new requirements placed on TV receivers or broadcasters. The success of this standard depends only on its perceived value and support within the industry.

In a separate action, the FCC seeks to update and improve upon the Emergency Broadcast System (EBS)[6]. The EIA has responded with comments to the Commission in this matter identifying the aspects of the EIA-608 that could be used as part of a more modern EBS system. It is unknown how the FCC will consider the capabilities of the EIA's EDS specifications in this matter.

EIA-608 CONTENT

The EIA-608 specification is intended to be the complete reference for all line-21 data services. The title of the specification is: LINE-21 DATA SERVICES FOR NTSC. It documents all aspects of line-21 captioning, text and EDS services for service providers, data encoders, and receivers. It covers the (minimum) required captioning services as established by FCC rules, as well as optional and enhanced captioning, text and data services.

The EIA-608 Specification includes the following sections:

- Signal Specifications for Line-21
- Field-Two Data Formats and Protocol
- R.P. for Caption Encoder Manufacturers
- Recommended Practices (R.P) for Caption Service Providers
- Extended Captioning Features
- R.P. for Caption Decoder Manufacturers
- R.P. for Text Mode
- Extended Data Service Packets
- EDS Encoding and Usage Issues
- EDS Encoder Design
- EDS Encoder Usage

EDS PACKET TYPES

The Extended Data Service packets are divided into *Classes*. These classes provide an organizational grouping of packets based on context. These classes also allow packet definitions to be reused in different contexts. There are six defined classes:

The *Current Class* includes all EDS packets that describe the current program.

The *Future Class* includes only EDS packets that describe future programs.

The *Channel Class* includes those packets that describe the source or channel.

The *Public Service Class* contains those packets relating to public service information such as Emergency Broadcast Services (EBS) or weather information.

The *Miscellaneous Class* includes packets relating to all other areas.

An *Undefined Class* is included to allow closed loop services to use the data channel in a way that will be ignored by standard decoders.

EDS FOR THE PROGRAM PRODUCERS

One of the first opportunities to benefit from the EDS system will be at the program production or post-production point. A number of caption provider organizations are making improvements to provide for the insertion of the new field-2 data services. When a program is produced, it may only be a small incremental cost to have basic EDS data added at the same point that captions are added. This level of EDS implementation promises to reach the largest possible audience for the lowest possible cost.

When EDS is included at the production stages, there will be no added generation loss. Encoding will be done simultaneous with standard line-21 captioning. The EDS will always remain a part of the program without any further action downstream. (In some cases downstream line-21 encoders will require a one-time modification to pass field-2 data.)

The largest group of EDS packets are best done during the production or post production phase. The *Program Name* and *Program Description* packets are the most obvious. Other packets such as *Length/Time-in-Show*, *Program Type*, *Audio Services*, *Aspect Ratio* and *Composite Packet-1* might can also be encoded during production.

These packets can then be carried all the way to the final viewer's TV receiver with no other participation or equipment needed in the video chain.

Even at this simplest level of EDS implementation, consumers and program producers can benefit. Consumers will benefit by being able to quickly identify many important aspects about the program when channel hopping. Program producers will benefit by better recognition of their program. With the use of the EDS packets, more interested viewers are likely to catch a program because they can recognize the program quickly by name. Programmers will be able to capture these viewers, even during commercial breaks, because of this program identification.

EDS FOR THE NETWORKS

The network level is the second level of opportunity for EDS. When EDS is implemented at the network level, it has the potential to impact programming nation wide. There can also be several levels of network participation.

The first and simplest level of participation is "passive-compatibility". This means the network has taken any steps necessary to ensure that programs previously encoded with field-2 data will pass unaltered. Usually only a periodic system check is all that will be needed so everyone receiving the signals can enjoy the benefits of the field-2 signals.

A higher level of network implementation would involve the creation and encoding of EDS (and other field-2 signals). In this case the network can benefit by being able to provide a large percentage of his programming with EDS. The network can also benefit by adding packets not included during production such as Network Name, and Program ID Number. These packets will benefit the network by creating a higher level of recognition by viewers for both the network and the program.

To the consumer, the inclusion of the *Network Name* packet gives him the ability to find the familiar programming sources when visiting other towns or when faced with a channel assignment shuffle by the local cable operator. This packet may also eliminate the need for the superimposed on-screen logos used by the networks by making them redundant.

The *Program ID* packet can be used to control consumer VCRs, assuring complete capture of programs even during unscheduled interruptions, delays or extensions.

EDS FOR THE LOCAL BROADCASTERS

The final point in the distribution of television programming to take advantage of the EDS capabilities is at the local broadcaster level. There are various levels of participation that can give unique benefits the local audiences.

Like the network level, the local broadcaster's first level of support for EDS comes at a passive level. Again the first goal will be to make any system updates that may be necessary to ensure the unaltered passage of the line-21, field-2 data signal. Once this has been accomplished, all upstream EDS will reach the local broadcaster's audience with no further attention needed at the local level.

Also like the network level, there are a number of areas for further participation by the local broadcaster. Only by the use of an active EDS encoder at the local broadcast level, can the full benefits of the EDS system be realized.

Local broadcasters can benefit a number of ways by adding these services. The most basic advantage will be in the competitive advantage for increased audience. The packets that should only be added at this level include the *Time-of-Day*, *Time Zone*, *Call Letters*, *Tape Delay* and all the *Public Service Class*.

In some cases, the local broadcaster has the most control over the scheduling of future programs, in this case it would be appropriate to encode all the *Future Class* packets at this level.

The *Time-of-Day*, and *Time Zone* packets can provide a very useful service to the consumer by providing an automatic clock setting feature. The *Call Letters* packet provides a way for the local broadcaster to identify his station directly on the viewers TV screen. This can benefit the consumer by allowing an easy method of finding his favorite stations. All of these packets will allow TVs and VCRs to achieve a more user-friendly, automated set-up process.

The packets of the *Public Service Class* can offer unique benefits for consumers not before possible. To achieve this service, the local broadcaster includes a data bridge to couple the National Weather Service's Weather Radio Specific Area Message Encoding (WRSAME) signals directly into the packetized EDS signal. This will allow EDS receivers to capture and display on-screen, any NWS warning messages automatically. These messages are encoded to identify the specific events, the local areas affected and the duration. These warnings can be accomplished without any interaction or attention by the local broadcaster and without interruption to programming or advertisements.

To the consumer, this advanced warning system has the potential to save lives and property. In addition it can be unobtrusive to unattended recordings. Because there will be less need to interrupt the program, viewers will be able to record programs during some of these events without capturing the usual interruptions by newscasters or weatherman.

Some local stations, that due to their format, don't or can't provide weather or news information, will be able to keep their viewers warned, without hiring additional staff.

CONCLUSION

The pending EIA standard for EDS creates opportunities for broadcasters and program producers to provide new services to their viewers. These new services promise to increase awareness, and provide easy identification of encoded programs. Several levels of support will allow a gradual penetration of EDS services to a point where full support for EDS is provided from production through distribution and the full benefits of the system can be achieved.

New EDS services can help achieve simpler and easier to use TVs and VCRs. New products with these capabilities are expected to be introduced in the near future. The first demonstration of broadcast and reception of EDS signals has been planned to coincide with the NAB Convention this year.

Because of the packetized nature of this EDS approach, it is expected to provide an easy migration path to future digital video transmission schemes including HDTV and compressed digital-525 formats.

REFERENCES

[1] EIA Publications, "DRAFT STANDARD EIA-608 Line 21 Data Services for NTSC", October 12, 1992.

[2] Kempter, Paul, "Closed Captioning Extended Data Services: Broadcaster Technical Implications" NAB Proceedings, 1993 (tentative).

[3] Title 47, Code of Federal Regulations §73.682, (22)(ii) "At times when Line-21 is not being used to transmit a program related data signal, data signals which are not program related may be transmitted, *Provided*: the same data format is used and the information is of a broadcast nature.

[4] 47-C.F.R., §73.699 (figure 17 B & C)

[5] Amendment To The Rules Relating To Permissible Uses Of The Vertical Blanking Interval Of Broadcast Television Signals. FCC MM Docket No. 92-305.

[6] Amendment of Part 73, Subpart G, of the Commission's Rules Regarding Emergency Broadcast System. FCC FO Docket No. 91-301 and FO Docket 91-171

CLOSED CAPTIONING EXTENDED DATA SERVICES: BROADCASTER TECHNICAL IMPLICATIONS

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The addition of Extended Data Services (EDS) to the Closed Captioning signal offers benefits for the broadcaster and all consumers beyond the current closed captioning functions. The business and data definition aspects of EDS encoding are discussed in a companion paper[1]. The technical implementation aspects are covered in this paper, and they include equipment needs, setup tasks, operations needs and encoder data handling theory.

INTRODUCTION

Pending FCC approval, line 21 field 2 of the NTSC television signal will contain captioning, text and EDS. In order to effectively implement EDS, it is important to understand how the various types of EDS data are encoded and the requirements of co-existing with captioning and text data.

Over 40 types of data could be encoded on the EDS data stream at various points in the television distribution system. The television distribution system is therefore described in terms of EDS data insertion points. Each point in the distribution system may add some EDS, captioning or text data and integrate locally originated and upstream (inserted earlier in the distribution process) data simultaneously. Although this data integration task is handled in the encoder, setup and operation must be handled correctly to preserve the integrity of all three types of data all the way to the television receiver.

First, an overview of the program distribution paths are detailed with respect to EDS data insertion points. This multi-level encoding concept allows pertinent data to be inserted at appropriate points in the program distribution path.

Next, the levels of participation by the various distribution points (program producers, networks, or stations) are detailed alongside the benefits. Each distribution point may choose to participate or not to participate, depending on the desired results at the consumers television receiver.

The last two sections give further insight about the encoder's internal operational tasks by defining the various data channels and their interrelationships. The basic method of integrating data packets from multiple sources at a typical distribution point are also explained.

PROGRAM DISTRIBUTION PATHS

The diagram in Figure 1 shows the possible distribution paths for a captioned program, with or without EDS data. The shaded boxes indicate potential re-encoding points that may add data packets to an existing captioning signal.

Each shaded area represents a potential re-encoding distribution point where a Closed Captioning signal may be present from the upstream source and will need to be integrated with locally generated data. For example, the show name and captions may be encoded on the post production master tape, the network name encoded at the network control center, and the local station call letters encoded at the local station. At the network control center and the local station, upstream captioning data must be captured, integrated with the locally inserted data (network name or station call letters), and re-inserted on line 21.

Program Distribution Path for Closed Captioned Programming

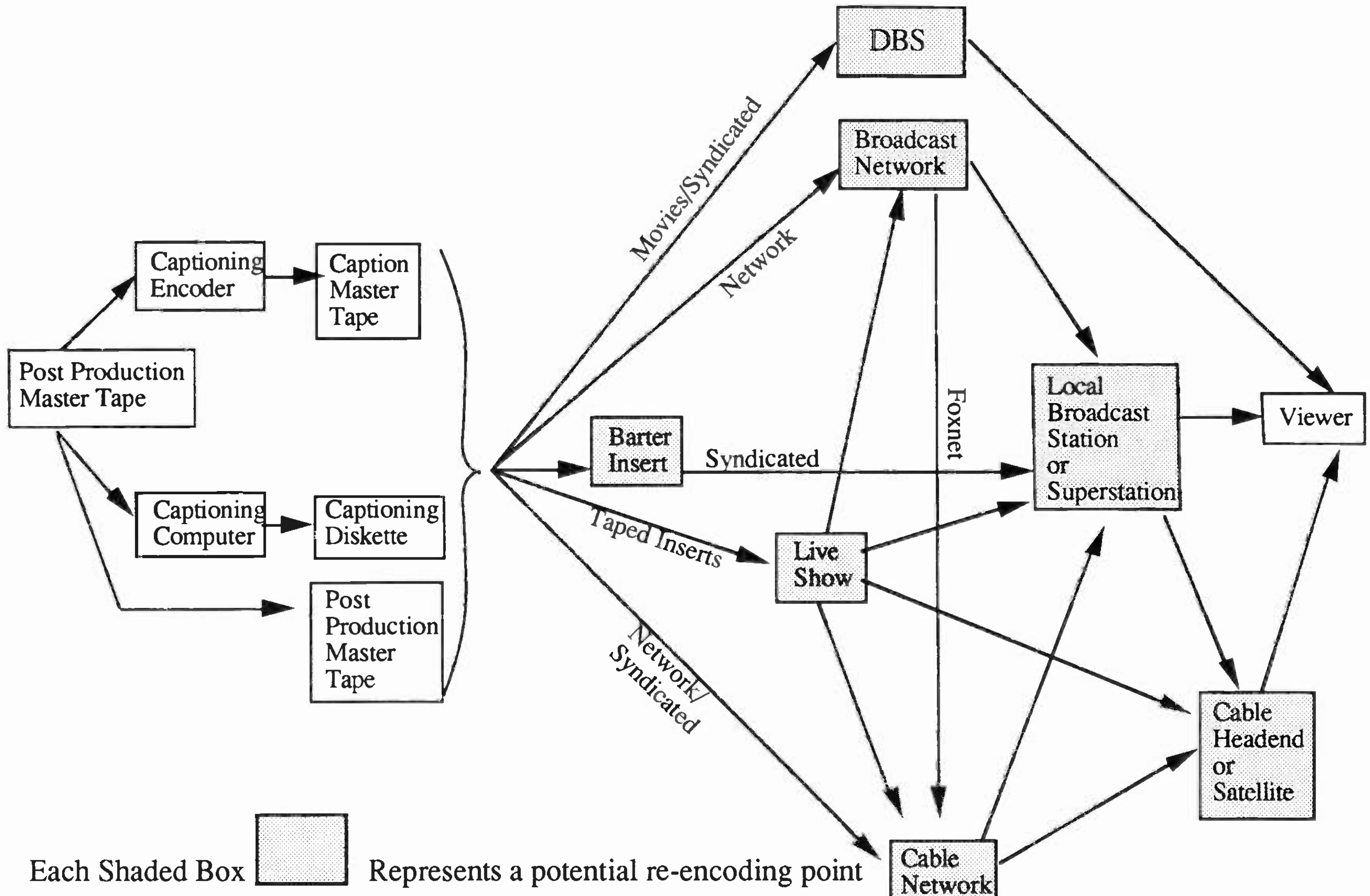

Figure 1. Distribution Paths

LEVELS OF PARTICIPATION

There are various levels of participation in encoding EDS. For example, a program producer may encode the captions and program name during post production and no further action or equipment is necessary at the network or station level for the program name to show up on the viewer's receiver. These levels of participation are summarized in Table 1 and further described here:

Program producer participation

If a program producer uses a caption service provider that has EDS encoding capability, then EDS data such as program name and time-in-show can easily be added as part of the captioning service.

Network participation

A network may choose to simply pass upstream captioning and EDS data or become further involved by inserting the network name. If the EDS data is not available from the upstream source (program producer), the network may choose to originate the current program name, program description, and network name at the network control center. The next level of participation involves inserting future network program information. Further participation can include Impulse Capture codes on promos.

Local station participation

The local station may also choose to simply pass upstream captioning and EDS data or become further involved by inserting the station call letters. If the EDS data is not available from the upstream source (program producer or network), the station may choose to originate the current program name, program description, and call letters. The next level of participation involves inserting future local program information. Further participation can include National Weather Service bulletins and Impulse Capture codes on promos.

LINE 21 DATA STRUCTURE

Details of the line 21 data structure are helpful in troubleshooting EDS problems. Figure 2 illustrates how data is structured in field 1 and field 2. Within field 1 are four data channels: CC1 (primary Synchronous Caption service), CC2 (special non-synchronous use captions), T1 (first text service), and T2 (second text service). Field 2 has five data channels: CC3 (secondary synchronous caption service), CC4 (special non-synchronous use captions), T3 (third text service), T4 (fourth text service), and EDS (Extended Data Services).

The Primary Synchronous Caption Service (CC1) is primary language captioning data that must be in sync with the sound, preferably to a specific frame. The Secondary Synchronous Caption Service (CC3) is an alternate captioning data channel usually used for second language captions.

The Special Non-synchronous channel (CC2, CC4) carries data that is intended to augment information carried in the program and need not be in sync with the sound. Delays of several seconds within the program are acceptable and would not affect the integrity of the data.

Text Services should use channels T1 and/or T2 if possible; T3 and T4 should be used only if T1 and T2 are not sufficient, and users should limit the combined data rate of T3 and T4 to 30% of field 2 capacity.

The EDS packets break down into several packet types (Current program information, Future Program Information, etc.). Composite packet types consist of multiple fields (Program Length, Title, Etc.).

Data channel bandwidth

Each field (field 1 and field 2) has a bandwidth of 60 7-bit characters per second. Each character consists of 7 data bits and 1 parity bit. Each encoding point has the capability to fill the entire bandwidth of a data channel preventing downstream encoders from adding new data without deleting existing data. The most likely example are Text services. Captioning data has top priority and therefore is not affected by the bandwidth used by Text and EDS services.

EDS data is less susceptible to decreased bandwidth as it passes through the distribution path due to it's repetitive nature. As the bandwidth decreases, the update or repetition rate of each EDS packet types decreases, but no

EDS Encoding Participant	Equipment required	Setup and operation required	Viewer Benefits
Program producer or Caption service provider	captioning/EDS encoder upgrade (output can be line 21 or data diskette)	Upgrade encoders to EDS capability	Program name, time-in-show, program description
Network	None	Allow line 21 to pass through network plant	If encoded at post production: Program information[1]
	1 Encoder per feed	Install encoder in program feed; one time encoder setup.	Network Name
		Install encoder in program feed; connect to network program lineup data source	If not provided upstream: Network Program information[1] Future network info[2]
	1 Encoder	Editing suite modification	Impulse capture for network show promos
Station	None	Allow line 21 to pass through station	All upstream network and post production EDS data
	1 Encoder	Install encoder in program feed; one time encoder setup.	Station call letters, time of day information[3], National Weather Service information
		Install encoder in program feed; connect to local program lineup data source	Current and Future Local program information[1,2]
	1 Encoder	Editing suite modification	Impulse capture for local show promos

[1]Program Information may include: Program Name, Length, Time-in-show, Program Type, Program Audience, Audio Services, Caption Services, Aspect Ratio, Program Description.

[2]Future program Information may include: Program Name, length, Program Type, Program Audience, Audio Services, Caption Services, Aspect Ratio, Program Description.

[3]Time of day information may include the local time, time zone, and tape delay packets.

Table 1 Impact of various levels of EDS encoding participation

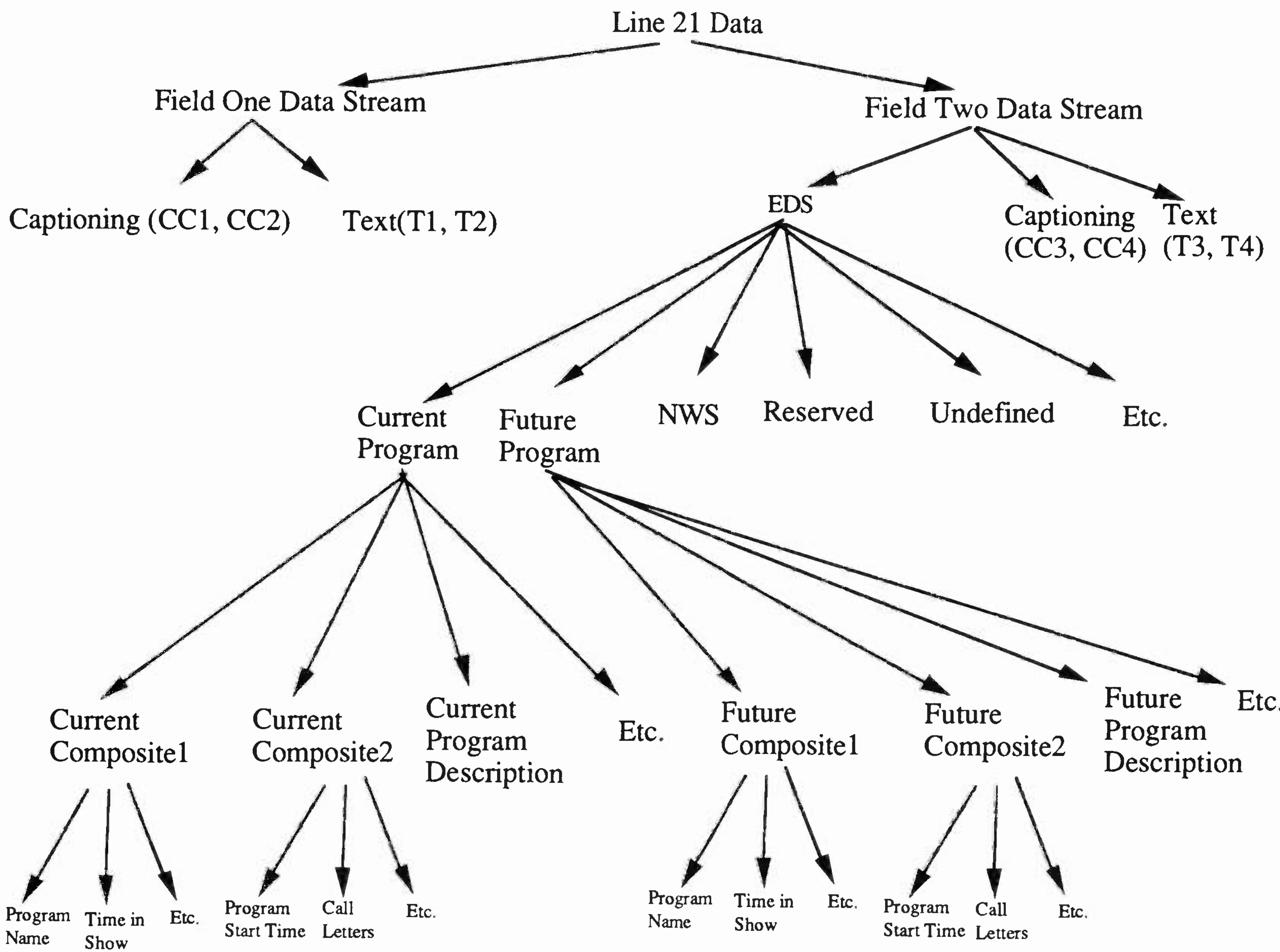

Figure 2. Line 21 Data Structures

data is lost. When the bandwidth available to EDS drops below 30%, the repetition rate will likely be slow enough to annoy the consumer who is channel grazing.

Therefore, text services on line 21 field 2 should never use more than 30% bandwidth to avoid significant impact on EDS services. If this guideline is exceeded, the text material may be significantly delayed or deleted at a downstream re-encoding point.

DATA PACKET INTEGRATION

Data packet integration occurs at each re-encoding point where local information is inserted into an existing captioning data stream. Data may come from several sources: upstream data (from line 21 of the incoming video), an in-house or remote communications link, a disk file (synchronized by SMPTE time code or internal clock), in-house video source, or pre-set data (such as network name or station call letters).

An example of a remote communications link source is a live show with captioning being supplied via modem from a captioning service provider. The captioning data will be integrated with the locally generated data (such as the network name or station call letters) at the origination point of the show.

Using a disk file as a captioning/EDS data source avoids creating another tape generation for non-live shows. The captioning data is captured on diskette and integrated with the locally generated data at the network or station.

EDS data timers

Each type of EDS data has a timer that tracks the amount of time left until the data is not longer valid and therefore should no longer be transmitted.

Timers ensure transmission of network program identification information during local commercials (when upstream program data is not being received). This ensures that the consumer, while channel grazing, can identify a show within a few seconds independent of what is being broadcast at the time. The proposed default values of these timers are in Table 2.

Packet Names	Initial Timer Value
National Weather Service (NWS) - Emergency	Set by valid period field in NWS packet
Impulse Capture	Length of Promo (entered by encoder operator)
Call Letters, Time of Day and Network Name	Forever (Enabled by operator)
All other Current and Future Class Packets	5 minutes but no more than time remaining in current show
Channel Information and Misc. Classes	5 minutes
Undefined Classes	Retransmit once within 10 seconds of receipt

Table 2 - EDS Initial Timer Values

CONCLUSION

Although EDS encoding may appear somewhat complex, most implementation complexities such as data packet integration are taken care of in the encoder. This simplifies the operational impact on the television program producers, networks, and stations.

The flexibility of the EDS encoding standard allows program producers, networks, and stations to independently choose from a variety of implementation levels. This permits a unique cost/benefit decision for each participant and avoids placing undue burden on other participants in the program distribution system.

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RFR MANAGEMENT

Wednesday, April 21, 1993

Moderator:
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ELECTROMAGNETIC RADIATION OF TOWER WORKERS: SAFETY STRATEGIES
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COMPLYING WITH THE FCC'S RADIATION HAZARD PROTECTION REQUIREMENTS DURING TOWER RENOVATIONS
Richard Mertz
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MEETING IEEE C95.1-1991 REQUIREMENTS
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RF PROTECTIVE CLOTHING FOR THE BROADCAST ENVIRONMENT
Donald T. Doty
Doty-Moore RF Services, Inc.
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TAKING MEANINGFUL RFR MEASUREMENTS WITHOUT ENDANGERING THE WORKER
David Baron
Holaday Industries, Inc.
Eden Prairie, MN

COMPLIANCE WITH LOCAL RADIO FREQUENCY RADIATION REGULATIONS
Gray Haertig
Haertig and Associates
Portland, Oregon
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ELECTROMAGNETIC RADIATION OF TOWER WORKERS: SAFETY STRATEGIES

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ABSTRACT

The protection of tower workers from potentially harmful radiofrequency electromagnetic radiation is an issue often overlooked. Yet it is an issue with which broadcasters are being confronted and which they must understand. This paper discusses the biological effects of RF radiation on personnel and the technical problems with determining safe levels, and presents practical and straightforward methods for solving those problems.

INTRODUCTION

For many years tower workers have been working on or near energized antennas. Until recent years little concern was shown for the biological effects on these workers of the electromagnetic energy emanating from those antennas. While it has long been known that harmful effects can be expected from high concentrations of microwave energy, the belief has prevailed that radio waves of the broadcast spectrum are not harmful. In the late 1970's, however, people began to question this assumption and a few people began to believe that they could associate instances of harm to the human body with RF radiation. After a good deal of research, in 1982 the American National Standards Institute released a standard called the "Radio Frequency Protection Guides[1]." The Federal Communications Commission has adopted these guidelines. Since January 1, 1986, the Commission has required that license applications (including renewal applications) contain either an environmental assessment which is to serve as the basis for further Commission action or a statement that the station operation will not have a significant environmental impact. *Such a statement, which all license applicants must make, is a sworn certification by the applicant that a determination has been made that humans will not be exposed to RF radiation in excess of the ANSI guidelines.*

The truth is, however, that the overwhelming majority of broadcasters have not made such a determination. They have simply checked off the block on their application form which, they know, will get their applications approved most quickly. The owners, operators, and engineers of some stations have not the slightest idea whether or not their transmitters, transmission lines, and antennas are radiating energy which exceeds the ANSI guidelines. Most have not even attempted to determine whether the general public is exposed to fields in excess of the standards, much less whether the station's employees are routinely exposed to such fields. And, until very recently, almost no one has addressed the issue of whether tower workers are being overexposed. It should be obvious that making a determination of whether or not the signal from an antenna at the top of a tower causes a high field area to exist at ground level near the base of the tower, implies nothing whatsoever about the safety of the space halfway or more up the tower. Yet license applicants routinely certify to the FCC that they have determined that no humans will be exposed to radiation in excess of the ANSI guidelines.

In August, 1992 the FCC issued a public notice demanding broadcaster compliance with its regulations and with the license representations. The notice specifically declares that

> "the licensee may not refuse to reduce the power on grounds that it could result in a temporary loss of audience or advertising revenue. Further, the licensee may not avoid complying with the ANSI guidelines even if a particular tower crew is willing to accept high RF exposure levels. . . . . Exposure of workers to RF radiation in excess of the ANSI guidelines and failure to comply with representations to the Commission in that regard are serious matters and may warrant further Commission action, including imposition of sanctions.[2]"

This paper will examine some of the biological and technical issues regarding the safety of tower workers. Some of the problems with ensuring worker safety will be discussed. Some specific and practical solutions will be proposed.

THE SOCIO-BIOLOGICAL PROBLEM

The health issues which concern tower workers range from simple headaches to death. The concerns are real. The exact dangers of exposure of humans to electromagnetic radiation are poorly known because of limitations in the data base of biological effects. The ANSI guidelines are admittedly conservative. But the concerns are real and they are justified. The fact that the data base is limited and the fact that the precise mechanisms by which health problems occur are poorly known is of no comfort to the tower worker who experiences health problems. Furthermore, when every man in an entire tower crew is suddenly afflicted with precisely the same malady, all with onset within a few minutes of one another, it stretches the imagination to presume that there is no local environmental cause. Such incidents do occur; they are not uncommon.

The health problems which have been associated with incidental RF exposure include, but are not limited to: headaches, dizziness, nausea, stomach aches, diarrhea, sterility, birth defects in offspring, leukemia, brain tumors, other types of cancer, and of course RF burns and electric shock. Some of these conditions are clearly known to be caused by radio frequency radiation exposure. The relationship of others is suspected but not well established.

The term *incidental* RF exposure is used to differentiate between exposure incurred in the course of working in, on, or near antennas and the exposure intentionally applied as a course of medical or similar treatment. These intentional exposures generally are carefully controlled; the effects of these controlled exposures are well known and well documented. The effects of incidental RF exposure are not well known or well documented because the circumstances of the exposure are not controlled, in either time or intensity. Indeed the person being exposed is often unaware that he has entered a region of high intensity RF energy. Complicating the issue is the fact that some conditions may not show any symptoms for months or years. When symptoms do occur, there may be little or nothing to link the condition to the long past radio frequency radiation exposure.

Also, the fact that tower workers have worked in high RF fields for so many years, generally without major complaint, is part of the problem of recognizing the hazard. This is partly attributable to the tower workers themselves. The stereotypical tower worker has somewhat of a *macho* personality. He is unlikely to complain to his fellow workers of minor illnesses or discomfort, lest he be ridiculed. The "old-timers" of the industry often disbelieve, and are quite vocal about their disbelief of, recent reports or research findings which are, admittedly, often tenuous and inconclusive. Competition also discourages some short-sighted tower workers from complaining about adverse effects. A small business tower worker may be very hesitant to demand or even to request concessions or safety accommodations from a station operator or engineer, feeling that, if he doesn't do the work as and when the customer wants it done, someone else will.

On the other hand the body of knowledge about the biological effects is contaminated by the illegitimate claims of a few workers trying to make workmen's compensation or other liability claims. In his effort to justify a claim, a worker may try to convince a court or other parties of a "clear" cause and effect relationship between a disease or injury and his tower work and exposure to RF radiation. In reality the relationship may be anything but clear and may not in fact exist at all. In today's litigious society every station owner, operator, and engineer needs to protect himself. Anything but the most conservative approach to the safety of tower workers is an opening to the threat of a lawsuit.

Certainly there are potential problems in exposing tower workers to excessive radiation, even though for many years the lack of data seemed to indicate that no problem existed. Clearly there are biological effects, even though it is less clear what the effects are or how they occur. So in discussing the issue of the safety of tower workers from radio frequency electromagnetic radiation exposure, the socio-biological problem is largely a problem of understanding and accepting that a potential problem exists, even though a large body of clear and incontrovertible evidence is lacking

THE TECHNICAL PROBLEM

In evaluating a region where a tower worker may go, we need to consider the sources of RF radiation on a tower and the coupling mechanisms by which the worker is exposed to RF energy. The radiation sources of concern in evaluating worker exposure may include: the radiating AM tower itself (in the case of an AM station), FM broadcast antennas, television broadcast antennas, two-way service antennas, paging service antennas, microwave dishes, and cellular antennas.

In establishing the requirements to report the evaluation of exposure potential, the FCC has categorically excluded broadcast auxiliary facilities such as studio-to-transmitter links (STL's), land mobile radio facilities, fixed communication services, amateur radio facilities, cellular radio, point-to-point microwave, and other relatively low-powered communications systems.[3] In their exclusion, however, the FCC noted that the Environmental Protection Agency questioned the exclusion of land mobile facilities. In order to avoid any potential liability, broadcasters may consider it prudent to include in their tower studies any emitters which have power levels above 100 watts and duty cycles above 50 percent. They may also wish to consider the effects of any microwave dish mounted and radiating in such a fashion that workers must work in the dish's main beam.

IEEE standard C95.1-1991[4] is expected to replace the ANSI standard. Its limits on electric and magnetic field and power density exposure in controlled environments are illustrated in figure 1. Its limits on induced and contact currents are shown in figure 2.

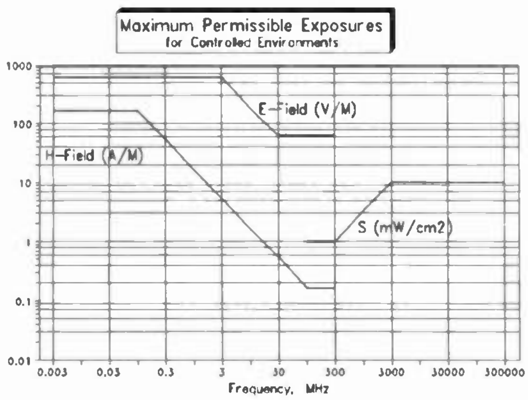

Figure 1. IEEE Standard exposure limits for electromagnetic fields in controlled environments.

The obvious RF radiation hazard on a tower is the potential for overexposure from directly radiated RF energy. But, in fact, workers are exposed to RF energy in a number of ways, including direct radiation, re-radiation, contact, and induction.

Re-radiation is a devious culprit. Many tower workers have experienced the phenomenon of a guy wire or a tower attachment or an adjacent antenna picking up energy from a radiating antenna and creating a local hot spot of re-radiation. This can obviously be a problem

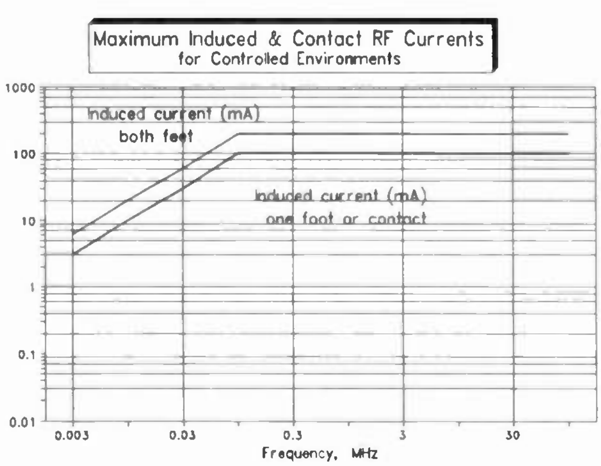

Figure 2. IEEE standard induced and contact radiofrequency current limits for controlled environments.

when the radiating emitter is a high power station. But a less obvious case of overexposure may occur during maintenance at a multi-station site or a multi-tower site. High RF field strengths can be present, even when the transmitter being worked on is completely shut down. An antenna for the station being worked on is likely to pick up high levels of energy from adjacent stations or towers. That energy, conducted via the transmission line to the final amplifier cabinet or the tuning unit being worked on, produces high field strengths in and around the tuning circuitry. The technician working on the supposedly "cold" circuitry may, in fact, be very significantly overexposed.

An individual who makes contact with a metallic object, which is at an RF potential different from adjacent objects, may suffer an electric shock or an RF burn. An example of this might be touching the insulated transmitting element of an FM or TV antenna. It can also happen when touching something mounted on the tower, permanently or temporarily, but not locally bonded to it. An example of this would be touching something being lifted into place by a steel cable on an AM tower. The steel cable, capacitively coupled to the tower but not bonded to it near the worker, can raise the piece being lifted to a very high potential. Significant arcing often occurs in this situation. A tower worker, holding the tower with one hand and reaching out to steady or to grab the piece being lifted with the other hand, receives an RF burn. This is not an uncommon situation.

The IEEE and ANSI standards limit the permissible direct contact current exposure. The exposure is clear and obvious when a worker receives an electric shock or an RF burn. But since overexposure often occurs at levels that are not clear and obvious to the worker, the broadcaster must prevent the possibility.

Induction exposure occurs in the eddy currents circulating within the tower climber's body and in the loop currents in the loop formed by the climber's body and the tower segment contacted by his hands and feet. Induction can be a significant hazard, particularly on AM towers. Research has shown that the eddy current contribution to the body's specific absorption rate is minimal, and that the eddy currents are not significant in evaluating hazards[5]. Loop currents, however, are another matter, and are very difficult to calculate. The human body's absorption is a complex function of frequency, tower height and wavelength, tower cross-sectional area, and, of course, power level. On an AM tower these loop currents, while very difficult to predict or calculate, may be high enough to overexpose a climber by a high factor.

Calculating intensities in the very near field region is a very complex problem. But the bottom line, in terms of worker exposure, is perhaps best summarized by Richard Tell:

> "Individuals climbing hot AM radio towers are subjected to strong electric and magnetic fields on the tower, which in most cases likely exceed the field strength limits of the ANSI RF protection guide. . . . . The contact current that can result when touching the tower of even 1 kW stations can easily exceed the contact current limit set in the recently revised ANSI RF protection guide (IEEE C95.1-1991) of 100 mA by up to a factor of 2½ times.[6]"

The best rule, and perhaps the only acceptable rule, for preventing the overexposure of tower workers climbing AM towers may be: "Turn it off."

SOLUTIONS

When tower workers bring to the attention of broadcasters the necessity of conformance with the ANSI standards, broadcasters usually reply with a response born of many years of experience. Typical responses include:

1. Work at night, after we're off the air.
2. Climb rapidly through the "hot" region and get above it before exceeding the 6-minute limit of the standard.
3. Work rapidly while the station reduces its output to a low power level for a short time.
4. Work rapidly without powering down, and complete the work before exceeding the 6-minute limit of the standard.
5. Use a Holaday meter to find "safe" areas.

Unfortunately each of these approaches has drawbacks.

Working at night is a feasible approach for some work. Tower climbers are very used to working at night when stations are off the air, or have one or more towers off the air, or are operating at reduced power. Lights can be changed. Transmission line jumpers can be changed. Small antennas like two-way sticks or even small dishes can be installed, deinstalled, or maintained. Many kinds of maintenance and service work can safely be done at night.

However there are many times when the work to be done is too complicated to be done safely in the dark. Making major modifications to a tower, such as replacement of structural members or adding poles and top hats, even raising and lowering of large, multi-bay FM and TV antennas, requires such a complexity of rigging and temporary bracing, that working in the dark brings its own safety hazards.

Any strategy which includes the words "hurry" or "rapidly" in its description, is inherently unsafe and is unacceptable. Tower work is by its very nature hazardous. Tower workers stay alive by taking their time and being careful. Anything which causes them to hurry is an open invitation to disaster.

Holaday meters and other instruments which attempt to measure the electromagnetic field intensity are good within their limits, but there are limits to their usefulness. They generally require a skilled user, both to operate the instruments and to interpret their results. Even the engineers supervising such measurements often cannot agree on their interpretations of the instruments' readings. Also the instruments require either an extra hand to hold or operate the unit or, if strapped to the climber's body, the instrument is often in the way. Most significant among the limitations of this technique is the fact that the instruments often fail to detect localized "hot spots" which can be significantly more hazardous than the surrounding area.

The ANSI standard and the IEEE standard address the issue in terms of electric and magnetic field strengths, power density, averaging times, whole and partial body exposures, etc. No attempt is made in these standards to consider radiation sources or safe distances from the sources. What the broadcaster needs to know is, "How far away from it does the man have to be, in order for me to be in compliance?" In order to ensure compliance with the ANSI or IEEE standards, when allowing people to work on or near a tower, each broadcast station must either turn off power or be able to certify that all locations in which personnel will be working are below the limits. His ability to make this certification comes with a tower profile.

A tower profile is the documented result of an engineering study which gives either measured or predicted values of both E- and H-field intensities or power density. If predicted values are used, the predictions must be made according to an accepted procedure. If measured values are used, the values must be recorded and maintained on file. If the tower profile shows that, with all antennas operating at full power, there are locations on the tower which are above the ANSI/IEEE limits, then the profile must indicate those areas and must dictate clearly that those areas must not be entered until power is reduced or turned off. If the broadcaster establishes a plan that permits workers to work in the high level areas after the power to one or more antennas is reduced, then a second tower profile must document the field values or power density at all locations on the tower at the reduced power levels.

A tower profile can be generated from the equations, tables, and graphs given in the FCC's OST Bulletin No. 65[7]. In this document broadcasters are provided with a means of determining where the ANSI limits are met and where they are exceeded. Predicted field intensities and power densities are derived as functions of distance from a radiating source. In the bulletin, the equations of section II or the tables and graphs of Appendix B can be used to determine the safe distance from an FM transmit antenna; the equations or the tables and graphs of Appendix C can be used for TV broadcast antennas. Unfortunately Appendix D, which addresses AM broadcast antennas, is limited to finding a safe distance from the tower for various power levels. This assists the broadcaster in finding the distance at which the general public can be considered safe. But obviously a worker on the tower is at zero distance, so Appendix D does not apply. It does nothing to determine where on a tower the surface fields are at a maximum or a minimum or to find their intensity. The information in Appendix D was generated using the Numerical Electromagnetic Code (NEC)[8]. This program is a highly accurate modeling tool in the far-field region, but is of very limited use in the close-in near field region of a worker on an AM tower.

As described in the OST Bulletin a conservative and acceptable approximation of the power density from an FM antenna is

$$S = \frac{360\ EIRP}{R^2} \qquad (1)$$

where

S = power density in milliwatts per square centimeter;

EIRP = effective isotropic radiated power in kilowatts; and

R = the distance in feet from the point being evaluated to the center of radiation.

(This equation is rearranged from the form given in OST Bulletin 65, to accommodate units more convenient to broadcasters.) A plot of this equation for various typical power levels of FM transmitters is shown in figure 3. This plot was generated using the graph generating capabilities of a computer spreadsheet. In using this information, care must be taken to ensure that that the power includes both the vertical and horizontal components, if using a circularly polarized antenna (a factor of 2) or an elliptically polarized antenna (a factor of between 1 and 2).

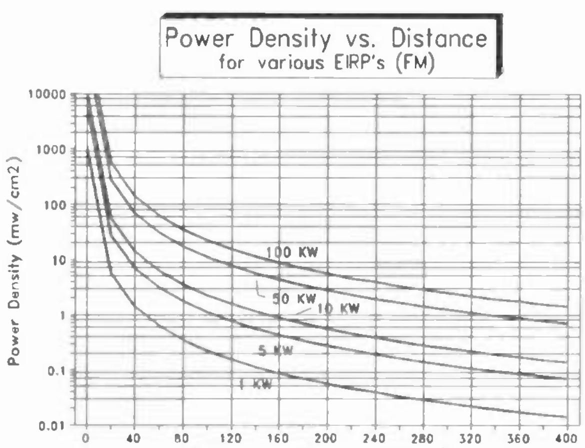

Figure 3. Power density vs. distance from FM transmit antennas at various power levels.

The equation for predicting power densities from television broadcast antennas is slightly more complex, because of the need to consider both aural and visual carriers, and the need to apply a relative field factor, F. The formula for determining power density in a direction downward from a television transmitting antenna is

$$S = \frac{360\ F^2\ [0.4\ VERP + AERP]}{R^2} \qquad (2)$$

where

S = the power density in milliwatts per square centimeter;

F = typical relative field factor in the downward direction;

VERP = total peak visual effective isotropic radiated power, in kilowatts;

AERP = peak aural effective isotropic radiated power, in kilowatts; and

R = the distance in feet from the point being evaluated to the center of radiation.

(Consult the OST bulletin for a complete discussion of the factor, F. A worst case approximation is to use a factor of F = 1.) A plot similar to that of figure 3 can easily be developed for a particular television transmitter, using equation 2 and a computer spreadsheet program with graphing capabilities.

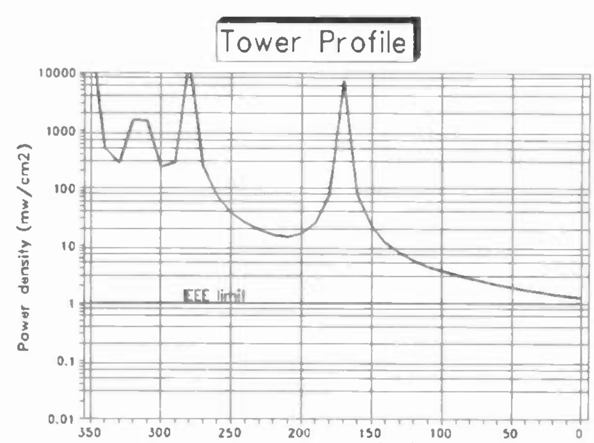

Figure 4. Example tower profile with all antennas at full power.

These equations can be applied to each emitter at every level on the tower to determine the total field or power density at each level. The spreadsheet program can then be used to plot a graph of total E-field and total H-field intensity or of total power density versus height on the tower. An example of such a profile is shown in figure 4. This example depicts a 350-foot tower with four stations operating as follows:

- TV Ch. 2, 100 KW visual, 22 KW aural, circularly polarized at 350 ft.
- TV Ch. 7, 100 KW visual, 20 KW aural, horizontally polarized at 280 ft.
- FM 88.1 MHz, 10 KW, horizontally polarized at 170 ft.
- FM 103.1 MHz, 50 KW, circularly polarized at 315 ft.

The tower profile in figure 4 clearly indicates that this tower is not to be climbed under these conditions.

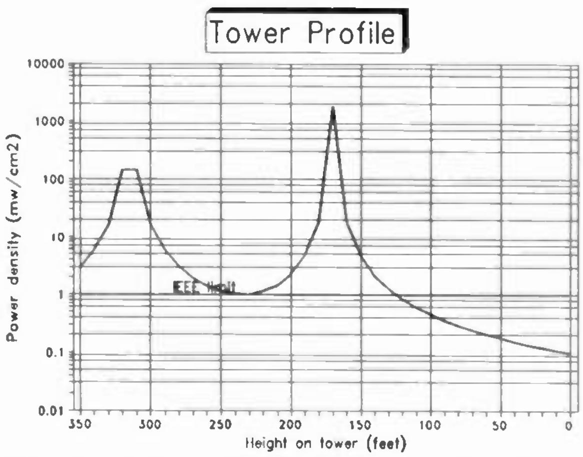

Figure 5. Example of tower profile with two emitters off the air and two at reduced power.

As an example of the change in the power density at various points on the tower, assume that the two television stations are off the air and that the FM stations have powered down to 2.5 KW and 5 KW, respectively. The profile of the tower in this condition is shown in figure 5, and it is quite clear where workers may climb and where they may not.

Tower profiles of this nature are not difficult to generate. These were done by simply applying equations (1) and (2) above. They can even be used dynamically with a desktop computer at a transmitter site. When major tower work must be done, which requires riggers to move about on a tower, a profile program can be used to quickly examine output power changes and predict where workers may go as a result of each change. Care must be used in setting up the program or spreadsheet to ensure that the equations are correctly applied and that assumptions made are valid. Such an effort should not be made at the last minute, while men are standing by, waiting to go aloft. But with proper forethought, planning, and preparation, a station's engineer can ensure that his tower can be climbed safely and with minimum disruption to operations.

Unfortunately, tower profiles of this nature do not work for AM towers. For an AM station a profile generally must be generated from measured data.

CONCLUSION

The Federal Communications Commission has put broadcasters on notice that they are all, jointly and severally, responsible for ensuring the safety of personnel working on and around their antennas, as well as the safety of the general public and other employees. Concerns for the safety of tower workers must include concern for compliance with the ANSI and IEEE standards for radiofrequency radiation protection. The practices of the past, including ignoring the standards and employing tower climbers who ignore the standards, will no longer be tolerated. Furthermore, the potential for lawsuits or other legal claims dictate prudence by the operators, owners, and engineers of all broadcast stations.

Yet the need for prudence and care need not dictate that broadcasters shut down their stations every time a beacon bulb needs to be changed. Careful engineering and foresight will minimize the disruptions required.

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COMPLYING WITH THE FCC'S RADIATION HAZARD PROTECTION REQUIREMENTS DURING TOWER RENOVATIONS

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ABSTRACT

The difficulty of modifying the structure of a tower with existing broadcast and two way tenants to accommodate a new broadcast tenant is, in itself, a challenge. Add to this challenge the FCC's requirement that licensees shall insure that tower workers are never exposed to radiofrequency radiation in excess of ANSI limits, and you have a project whose difficulty is increased an order of magnitude. This paper will explore such a project and the way in which the exposure aspect was monitored and controlled.

INTRODUCTION

When a station renews its license, it must certify that the transmitter site does not expose the general public, station employees, or tower workers to RF radiation levels that exceed the ANSI guidelines adopted by the FCC. This certification should come only after the station has fully reviewed its operation for compliance. In most cases the general public and station employees are protected simply by staying some distance from the tower. However, if it becomes necessary to do repair or installation work on the tower, just how do you insure that the tower crew is not exposed to RF levels in excess of ANSI limits? Remember, a tower worker cannot release a station from this responsibility.

THE PROJECT

I recently completed a project that required structural modification be made to an existing tower in order to prepare for the installation of an additional FM station's antenna. The tower already had a class B FM station, a 5 kW daytime AM station, and numerous two way tenants. Since the existing tenants did not have alternate transmitter sites from which they could operate during the tower renovations, it was necessary to devise a plan that would permit the changes while keeping the present tenants on the air.

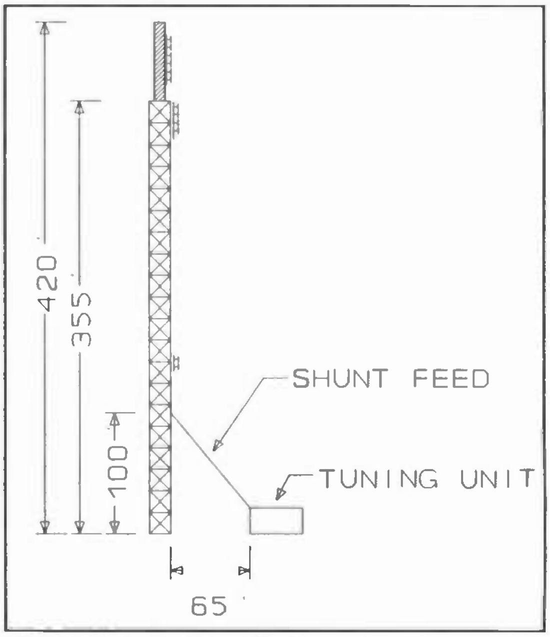

Figure 1- Tower installation prior to modifications

Figure 1 shows the tower and tenants before any renovations. The Class B FM station, which I will call station WFFF, had three antennas on the tower. Mounted on the pole is a 6 bay half wave spaced antenna, which is operated at an effective radiated power ("ERP") of 41 kW.

The second FM antenna is WFFF's auxiliary antenna mounted directly below the 6 bay pole mounted antenna. WFFF also has an emergency 2 bay antenna mounted about 110 feet above the ground.

The AM station, which I will call WAAA, is a 5 kW daytime station that is shunt fed at a point about 100 feet above the base of the tower. The tower is grounded at the base by four 6" ground straps. The tower also supports numerous two way antennas.

The required renovations include the installation of an additional pole, structural changes to the top 20 feet of the tower, and the installation of cross bracing from a point 311 feet to 336 feet above ground level. Figure 1 shows that WFFF's main and auxiliary antennas would have to be removed to permit the tower crew to make the tower renovations. Prolonged operation on the emergency 2 bay antenna would be limited due to severe reduction in station coverage. The power to this antenna needs to be reduced to insure a safe working area around the base of the tower.

Clearly the top two FM antennas would have to be removed to permit the tower crew access to those areas of the tower that needed modifications. In order to keep the FM station operating with reasonable coverage it was decided that the 4 bay auxiliary antenna would be relocated to a clear area in the center of the tower (Figure 2). This would permit a safe work area near the top of the tower as well as around the base of the tower.

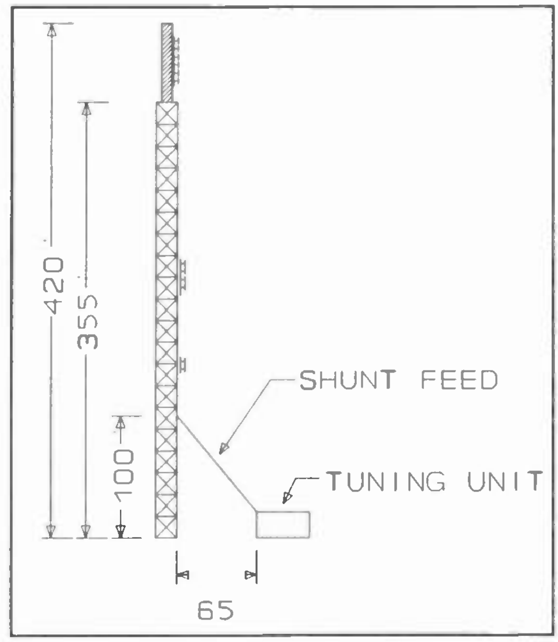

Figure 2-Relocation of 4 bay auxiliary antenna to the center of the tower.

The two bay antenna was left at 110 feet above ground. This antenna would be used whenever lift chairs were used to carry the crew to the top of the tower. Additionally, this lower antenna would be used when workers had to come close to the mid-tower four bay antenna.

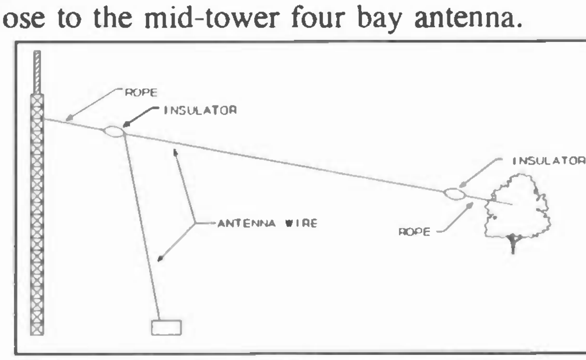

Figure 3-Temporary AM antenna installation

To keep WAAA operating during the renovation, it would be necessary to construct an AM antenna away from the tower. The tower would not be used as an AM radiator during the entire renovation. Figures 1 and 2 show that the tuning network is approximately 65 feet from the base of the tower. Using about 300 feet of phosphor bronze wire, insulators, and heavy rope, a temporary antenna was constructed as shown in Figure 3.

The temporary AM antenna was secured to the tower by about 50 feet of rope. Note that the rope was not tied to the top of the tower. This section was to be removed and rebuilt later in the project. The antenna was folded about 200 feet above ground and the loose end tied, using another length of rope, to a convenient tree. The vertical part of the wire antenna was connected to the tuning network. The wire was held mechanically taut using a drum partially filled with rocks and gravel.

PREPARATION

Once the long wire antenna was in operation, measurements were taken using a Holaday meter. The results of the measurements show that at 5 kW, the electric field was approximately half the ANSI requirement of 632 V/m. at a distance of 12 feet from the wire. This area was marked off with stakes and red warning tape. The crew was instructed that they were not to venture within this marked area. The magnetic field measured 0.19 A/m which is about 12% of the ANSI limit.

Next the fields around the tower were measured. The magnetic field was not measurable. The electric field was about 50 V/m, which is 7.9% of ANSI limit. Theses measurements were take while WFFF was using its main antenna. Since the antenna elements are spaced a half wavelength apart and its radiation center is 365 feet from workers on the ground, its contribution to the total exposure was small. Another series of measurements were made with WAAA operating at 3 kW. This reduced the measured field around the tower to 25 V/m or 4% of the ANSI limit. Hot spot measurements made 8 cm from the tower members read even smaller. Electric fields were measured about half way between the tower and the long wire antenna operating at 3 kW. This is the approximate location of the winch. The resulting measurement of 100 V/m is 15.8% of the ANSI limit.

Total exposure to the workers is the sum total of contribution from each RF source. From the measurements, we know that at 3 kW the contribution from WAAA is about 15.8%. There are various two way users. Even though the FCC excludes two way radios from RF exposure requirements, for this project 5% of ANSI limit contribution was assumed due to the relatively low powers and short duration of operation. This site uses master land mobile antennas for transmit and separate master antennas for receive. As an additional safety precaution the crew was told to avoid the transmitting antennas.

Adjacent to this tower is an AT&T microwave site. This is a standby facility and is not in use. Workers at the AT&T site verified that the facility was off. Also, approximately 1600 feet south of the subject tower is a non-commercial FM station transmitting site operating at an ERP of 75 kW. Using formula 4 in OST Bulletin no. 65[1], it was determined that the contribution from this RF source is 2.1%.

WFFF's contribution was calculated using formula 4 from FCC OST Bulletin No. 65[2].

$$S = \frac{(0.64)\,EIRP}{\pi R^2} \qquad \textbf{(1)}$$

Where:
S = Power Density
EIRP = Equivalent Isotropic Radiated Power
R = Distance to the center of radiation in meters

To put this formula to work, the station's

ERP is multiplied by 1.64. This converts ERP to EIRP. Since the FCC uses ERP, which is relative to a half wave dipole antenna, the 1.64 figure is the conversion factor for the gain of a half wave dipole relative to an isotropic radiator. Also the ERP is the sum of the power in both the vertical and horizontal planes. Since all the FM antennas used on this project are circularly polarized, we simply multiplied the nominal ERP times 2.

Substituting, these values

$$S = \frac{(0.64)(1.64)(ERP\times 2)(1000\frac{mW}{W})}{\pi(R(100\frac{cm}{m}))^2} \quad (2)$$

where:

S = Power Density in mW/cm²
ERP = Effective Radiated Power in Watts
R = Distance to the center of radiation in Meters

Using this formula, the radiated contribution to the total exposure can be calculated. WFFF's contribution from the top mounted 6 bay half wave antenna is calculated as follows. Please note that this is a worst case situation and does not take into account the half wave antenna's vertical radiation pattern. It is assumed that the field is radiated equally in all directions. The antenna's center of radiation is 375 feet above ground. Converting this to metric yields a distance of 114.3 meters. Two meters is subtracted to compensate for a person approximately 2 meters in height. Therefore, the value of R equals 112.3 meters. The station's ERP is 41 kW or 41,000 watts.

Substituting these values:

$$S = \frac{(0.64)(1.64)(82{,}000)(1000)}{\pi(11230)^2} \quad (3)$$

thus

$$S = 0.217\frac{mW}{cm^2} \quad (4)$$

expressed as a percentage of the ANSI limit:

$$\frac{0.217\frac{mW}{cm^2}}{1.0\frac{mW}{cm^2}}\times 100 = 21.7\% \quad (5)$$

summing all the contributing sources:

WAAA	15.8%
WFFF	21.7%
Two Ways	5.0%
Non-Commercial FM	2.1%
Total Exposure	44.6% of ANSI limit

From this information, it was determined that the workers could work on the ground and not be exposed to radiated fields in excess of ANSI limits.

The next problem was to devise a method to determine to what height the crew could climb on the tower and still be under the ANSI limit. First, it was necessary to determine which antennas would be used, when, and what ERP each antenna could radiate. In order to determine the ERP of each antenna and the transmitter power output (TPO) required, the following data was collected.

Antenna	Antenna Gain	Coaxial Cable Efficiency
6 Bay	1.9030	87.5%
4 Bay	2.1332	92.5%
2 Bay	0.9978	95.8%

Using this data in formula 6, it was easy to determine the ERP for the each antenna.

$$ERP = TPO \times G \times E \qquad (6)$$

where

TPO =Transmitter Power Output in Watts
G =Antenna Gain
E =Transmission Line Efficiency

Next, determine where the workers will be on the tower and measure the distance from that point to the center of radiation. Now substitute the result from formula 6 and the distance into formula 2. The result is the exposure in mW/cm² for station WFFF. This result, expressed as a percentage of the ANSI limit, is added to the percentages from the other sources.

For example, the crew needs to do work on the tower at 300 feet above ground. At this point they are 75 feet away from the top mounted 6 bay antenna. This means that the crew can climb past the two bay and 4 bay antennas without any power. The power to the 6 bay antenna would have to be reduced. Since the work would be done at 300 feet, a buffer zone of 10 feet was added to this height. The crew was told that 310 feet was their maximum height.

We know that all other radiating sources comprise 22.9% of the ANSI limit. We now need to determine what transmitter power would be required to make up the remaining 77.1%. As a note, even though 77.1% was the maximum percentage of the ANSI limit that WFFF could operate, for this project we would use a figure somewhat less. In most cases WFFF was run at about 60% of the ANSI limit. Using formula 6 and then formula 2, the exposure for a given TPO could be calculated. In this case with the top of the work area being 65 feet from the center of radiation, a TPO of 2100 watts yielded an ERP of 3500 watts. The power density at 310 feet above ground was .596 mW/cm² or 59.6% of the ANSI limit. This added to the other 22.9% put the exposure at 310 feet at 82.5% of the ANSI limit. Since the crew was working at 300 feet, the actual exposure was somewhat less than the calculated value. Keep in mind this method does not take into account the vertical radiation pattern of the 6 bay half wave spaced antenna.

GETTING STARTED

It may have seemed more advantageous, especially from the WAAA's and WFFF's managements' point of view, for all work to be performed during the overnight off air hours. While, from their point of view, this work is a nuisance and requires them to operate at reduced power, it is also extremely dangerous for the crew, surrounding property owners, and workers on the ground to rig and set tower sections, poles, antennas, and coaxial cables in the dark. The tower crew and I took the position that only work that required the crew to be close to several of the antennas at one time or work that required all stations to be totally off, would be done in the dark.

With the 4 bay antenna mounted in the center of the tower, work could begin to remove the top antenna and its mounting pole. Before starting, a written set of directions was prepared listing the power levels for WAAA and WFFF. These directions listed the power levels the power levels required while the crew is climbing. It was arranged, using the various antennas and power levels, that the crew never was within the aperture of a powered antenna. To do this power from the topmost antenna was reduced so the crew could climb or be lifted to a safe point halfway between antennas. Once they reached this point, they waited until the top antenna was turned off and a lower antenna was turned on at reduced power. One this was done, the crew continued their ascent. Once at the work site, the crew was given a lower limit below which they could not descend. The power was raised on the lower antenna to a level that would not exceed the ANSI requirements at the lower limit given to the crew. Each member of the crew had his own walkie-talkie and was aware of the lower limit. As a practice, if any crew member needed to go below this limit for any reason, they had to wait until the exposure was calculated and the power was lowered to the required level.

Here is an example of the type of written directions that should be followed by all concerned. The percentages in parentheses are WFFF's transmitter power output meter reading.

1. Tower Crew on Ground

A. WFFF
1. on top antenna at 24.5 kW TPO (100%)
2. on middle antenna at 20 kW TPO (81.6%)
3. on bottom antenna at 10 kW TPO (40.8%)

B. WAAA
1. 5 kW

2. Tower Crew Climbing (assumes top antenna is functional)

A. WFFF
1. Lowers power to 5 kW TPO (20.4%) until crew reaches 295 feet above ground level.
2. When crew reaches 295 feet above ground, WFFF switches to middle antenna at 3 kW TPO (12.2%).
3. When crew climbs above 330 feet, WFFF raises power middle antenna to 5 kW TPO (20.4%)

B. WAAA
1. Lowers power to 2 to 2.5 kW

3. Tower Crew Descending (assumes top antenna is functional)

A. WFFF
1. When crew climbs below 330 feet, WFFF lowers power on middle antenna to 3 kW (12.2%)
2. When crew reaches 295 above ground level WFFF switches to top antenna at 5 kW TPO (20.4%)
3. When crew reaches 112 feet above ground, WFFF raises power to 24.5 kW TPO (100%)

B. WAAA
1. When crew reaches ground WAAA raises power to 5 kW

4. Tower Crew Climbing (assumes top antenna is NOT functional)

A. WFFF
1. Lowers power on middle antenna from 20 kW TPO to 1 kW (5 %) while crew climbs to 164 feet above ground.
2. When crew reaches 164 feet above ground, WFFF switches to lower antenna at 1 kw TPO (5%)
3. When crew reaches 295 feet above ground, WFFF switches to middle antenna at 3 kW (12.2%) TPO.
4. When crew climbs above 330 feet, WFFF raises power middle antenna to 5 kW (20.4%)

B. WAAA
1. Lowers power to 2 to 2.5 kW

5. Tower Crew Descending (assumes top antenna is NOT functional)

A. WFFF
1. When crew climbs below 330 feet, WFFF lowers power on the middle antenna to 3 kW (12.2%)
2. When crew reaches 295 feet above ground, WFFF switches to lower antenna at 1 kW (5%) TPO.
3. When crew reaches 164 feet above

ground, WFFF switches to middle antenna at 1 kw TPO (5%)
4. When crew reaches ground WFFF raises power to 20 kW TPO (20.4%)

B. WAAA
1. When crew reaches ground WAAA raises power to 5 kW

During any project that requires workers be on a tower, both the station engineer and crew foreman MUST be aware of each crew member's position on the tower! NO EXCUSES! Safety is the top priority on any project.

PROBLEMS

As you can see from the instructions, movement up and down the tower was reasonably simple and in most cases worked. However, many situations arose that were out of the norm and required an on site evaluation. Each situation was evaluated based on where the workers would be in relationship to the active FM antenna. To simplify the calculations, I entered two programs into a programmable calculator. The first program used the antenna gain and coaxial cable data to determine what TPO (transmitter power output) would be needed to produce a desired ERP. The second program calculated the exposure in mW/cm^2 given the distance in feet and the ERP in watts. The same guidelines as stated above were used when determining the total exposure.

A problem that was not anticipated during the planning of this project was WAAA's signal, even at reduced power, was induced into the winch cable. Past experience, gleaned from previous site engineers, never described the degree of arcing that occurred. The floating winch cable and "headache" ball created an unacceptable "hot spot." Several solutions were tried, including 21 kV lineman's insulated gloves. The solution that worked best is shown in figure 4.

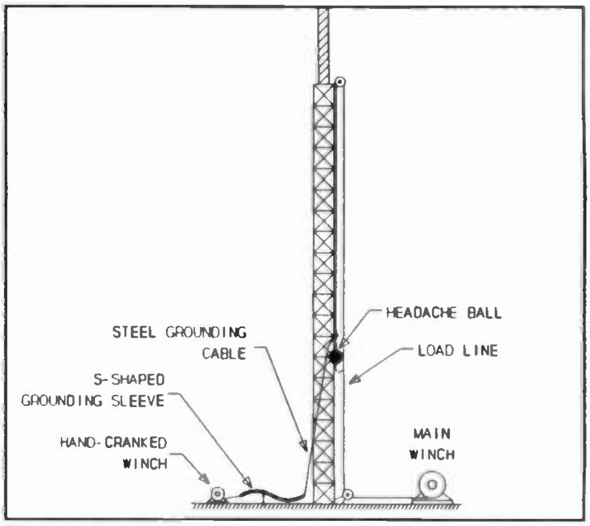

Figure 4 - Solution that reduce AM field induced in the winch cable.

A steel cable from a hand operated winch was passed through a grounding sleeve. The grounding sleeve was nothing more that a piece of 7/8" air dielectric heliax with the center conductor and insulation removed. A piece of the outer jacket was removed in the center for connection to the station ground. This grounding sleeve was connected to the transverse strap between the AM station's tuning unit and the tower. The coax was bent into an "S" shape and was a very efficient grounding device. The end of the steel cable was bolted to the winch cable above the "headache" ball using a "Crosby Clip." The reel was not grounded because current through the wire on the reel would have set up the potential for a strong magnetic field. Measurements confirmed that no magnetic field emanated from the reel it and was safe to operate by hand. The Holaday meter did not register a hot spot on the "headache" ball or hook.

This grounded winch cable, while safe for the tower crew, created a problem for WAAA. Each time the ground wire and winch cable combination was move more than a few feet from the tower the tuning of the station's transmitter would change. It was necessary to constantly adjust the tuning AM transmitter during operation of the winch.

On top of this problem, there were periodic jumps in the wire antenna's base impedance. A jump of almost 20 ohms! After a great deal of investigation we found the problem. During the rebuilding of the top twenty feet of tower, it was necessary to install temporary guy wires. When these guy wires were between uses they were simply tied off to a convenient fence. We thought that since these guy wires were sectionalized for use on an AM tower that simply keeping them away from the tensioned wires was sufficient. It was necessary, as we later found, that a fairly large space between the temporary guys and the tensioned guys was necessary to prevent this problem.

CONCLUSION

In order for a project of this magnitude to be successful and safe, it is of utmost importance that all parties involved understand their responsibility. If you are a station operating on a tower used by other stations and/or two way users, it is your responsibility to insure that tower crews and service personnel are protected from being exposed to radiation limits in excess of the ANSI/FCC required limits.

During the course of a project, such as the one described here, problems will arise that put all tenants at a disadvantage. Most station engineering personnel understand this. However, it was very difficult for station management.

On August 19, 1992, the Mass Media Bureau of the FCC issued a public notice reinforcing the Commission position on radiation hazard protection. They stated:

"It has come to our attention that some licensees either may not understand their responsibilities or may not be diligent in protecting humans from excessive RF radiation, particularly in cases where maintenance and repair work must be performed on or near antennas, tuning elements and transmitters. The obligations to protect humans form excessive RF radiation does not permit any exceptions. If for example, it is necessary that a tower crew work on or near an antenna, the power to the antenna must be reduced for as much and as long as necessary to avoid exposing the tower crew to RF radiation in excess of the ANSI guidelines. The licensee may not refuse to reduce the power on the grounds that it could result in a temporary loss of audience or advertising revenue. Further, the licensee may not avoid complying with the ANSI guidelines even if a particular tower crew is willing to accept high RF exposure levels.

We recognize that multiple radiators located on the same or nearby towers present a special problem. Nonetheless, all licensees are jointly responsible for complying with the ANSI guidelines and, therefore, must coordinate their maintenance and repair activities and take any other appropriate steps necessary to ensure that no humans are exposed to radiation in excess of the recommended limits."[3]

If you are a tenant at a multiple user antenna site, as much as your management dislikes the idea of reduced power operation, the station is still responsible regardless of which tenant is doing the work on the tower. If you are an FM station, I recommend you keep an auxiliary site available for use. In many cases coverage from an auxiliary site is better than extremely low power operation from the main site. It beats being off the air when the top beacon lamps need replacing.

If you are a user on a multiple user tower, all broadcast tenants should formally put in writing a plan by which the ANSI guidelines can enforced. This plan should be available at the site and should be submitted with the next license renewal as proof of your station's diligence in this matter.

ACKNOWLEDGEMENTS

I want to thank Gene Cummins from United States Tower Services, Inc., who helped me with the drawings shown in this paper. I also must give credit to Tom Jones of Carl T. Jones Corp. who came up with the ground wire idea and Tom Bunk of US Tower for the "S" shaped grounding sleeve.

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MEETING IEEE C95.1-1991 REQUIREMENTS

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Abstract

The new IEEE C95.1-1991 Standard limiting human exposure to RF energy is more complex than the present ANSI C95.1-1982 Standard. Although it introduces some new provisions that will complicate matters, it does contain certain provisions that may benefit broadcasters when the Standard is adopted by the FCC. This paper analyzes both the positive and negative aspects of the 1991 IEEE Standard and provides specific guidance for broadcasters trying to ensure that they continue to comply with appropriate regulations in this area of high liability.

HISTORY

The American National Standards Institute (ANSI) is an industry-supported organization that sets voluntary consensus standards for a variety of products and conditions. Several ANSI standards directly affect broadcasters, including those standards that concern buildings and other structures [1], electrical performance of television broadcast transmitters [2], and exposures to radiofrequency radiation [3]. Since ANSI is not a regulatory agency, it cannot enforce compliance with its standards. Rather, governmental agencies may use the ANSI standards as a basis for their own standards that will be enforced by those agencies. Such is the case with ANSI Standard C95.1-1982, which established guidelines for human exposure to electromagnetic fields. This voluntary standard was adopted in its entirety by the FCC, effective January 1, 1986. The FCC enforces compliance with this standard and, accordingly, requires statements concerning compliance with this standard on all applications for new or modified broadcast facilities.

At five-year intervals, ANSI reassesses and revises its standards as necessary, so the ANSI C95.1-1982 Standard was due for reaffirmation or revision in 1987. The process was indeed begun by ANSI, but it was handed off to the Institute of Electrical and Electronics Engineers (IEEE) for completion. In 1991, the IEEE completed its revision of the ANSI Standard and formally adopted the revision as IEEE Standard C95.1-1991. In turn, ANSI adopted the IEEE's revised standard as its own on November 18, 1992, calling it "ANSI/IEEE C95.1-1992."

This new standard was to succeed ANSI C95.1-1982. However, at the time of this writing, ANSI's adoption of the IEEE Standard has been stayed, in response to an appeal by the author's firm. The appeal alleges lack of consensus and procedural problems in the face of unaddressed technical concerns. Therefore, references to "ANSI" in this paper will be to the 1982 Standard, while the references to "IEEE" will be to the 1991 Standard.

In spite of the delay currently at hand, it is certain that some form of the IEEE Standard will be adopted by ANSI. Then, within a year of ANSI's official adoption of the new standard, the FCC is expected to revise its rules to incorporate the new standard. Beginning at that time, presumably, broadcasters will responsible for compliance with the new limits and conditions.

SUMMARY OF CHANGES

The IEEE Standard incorporates changes from the ANSI Standard in four major areas:

- Standards Coordinating Committee No. 28 (SCC28), the IEEE group tasked with updating the ANSI Standard, justified an additional safety factor in certain situations, although no new studies had been published to indicate adverse cumulative effects of low-level exposures. Thus came the biggest change from ANSI: new "uncontrolled" (*i.e.*, public) exposure guidelines, generally established at one-fifth of the "controlled" (*i.e.*, occupational) exposure guidelines.

- Also for the first time, guidelines were included for body currents; examination of the electric and magnetic fields will no longer be sufficient to determine compliance.
- The occupational guidelines were mostly unchanged, but considerable minor adjustments have been made, featuring the relaxation of the guidelines at certain frequencies and the introduction of breakpoints at new frequencies.
- Finally, the measurement procedures were changed in several aspects, most notably with respect to spatial averaging and to minimum separation from reradiating objects; the consequence of these changes is further effective relaxation of the exposure guidelines, particularly for public exposure conditions.

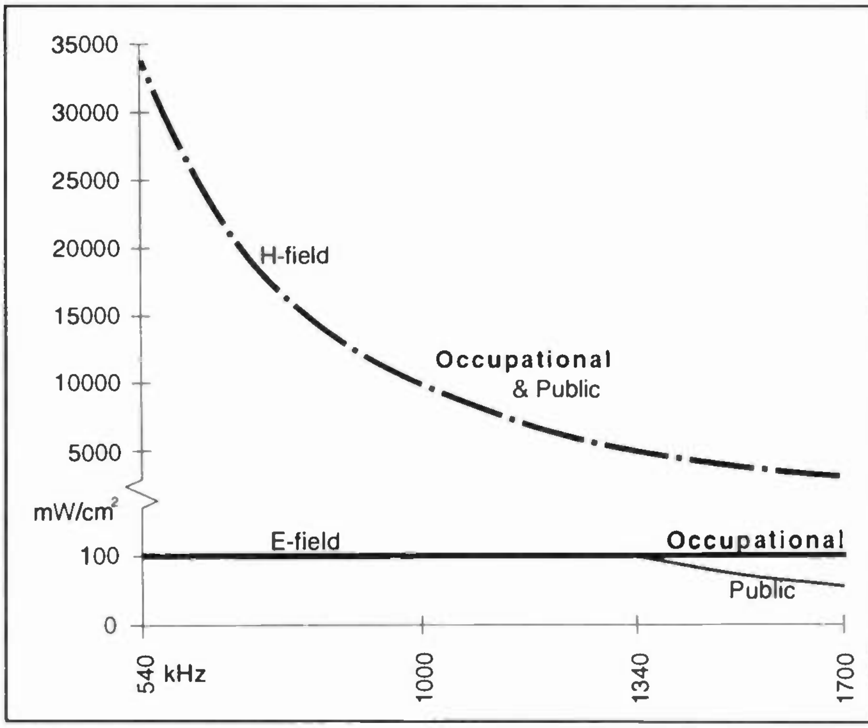

Figure A. IEEE guidelines for AM frequencies.

E-FIELD & H-FIELD GUIDELINES

AM Frequencies. Figure A above shows the IEEE exposure guidelines for electric and magnetic fields (E- and H-fields) in the AM band. Note that only the occupational E-field limit is constant across the band, remaining equal to the ANSI level of 100 mW/cm^2. For frequencies above 1340 kHz, the new public E-field guideline appears, dropping with the square of the frequency to its most restrictive level of 62 mW/cm^2 at 1700 kHz. The new IEEE H-field limits also decrease with the square of the frequency, and they are greatly relaxed from ANSI, by a factor of 340 (25 dB!) at the bottom of the band to a factor of 34 at 1700 kHz. There is no difference between the public and occupational guidelines for AM H-fields.

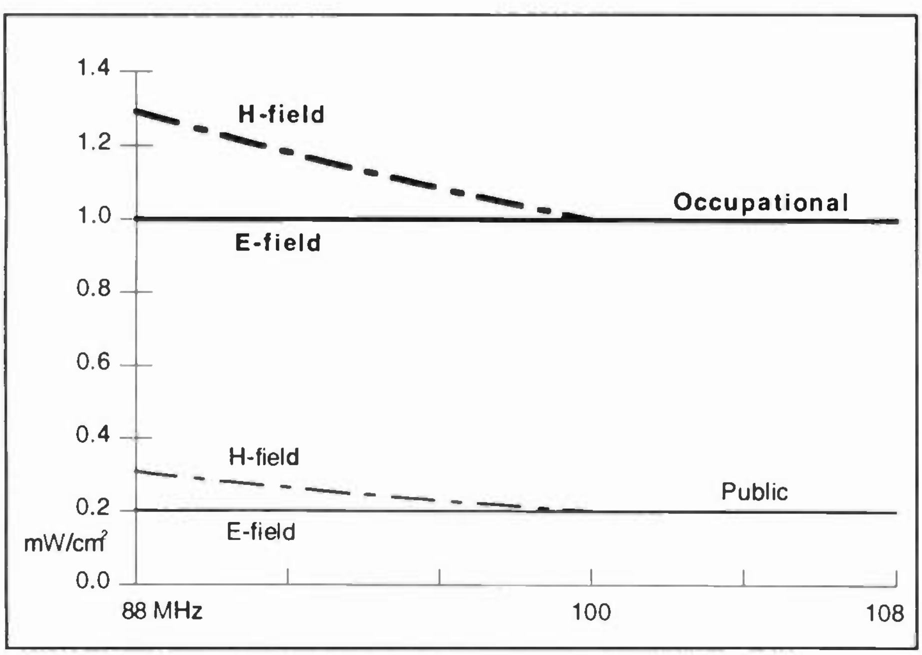

Figure B. IEEE guidelines for FM frequencies.

FM Frequencies. Figure B to the left shows the IEEE exposure guidelines for the 20 MHz spectrum block assigned to FM stations. The occupational E-field level of 1 mW/cm^2 is unchanged, and the new public E-field level is at 0.2 mW/cm^2, five times lower. (This is the "200 microwatt" level that has been commonly mentioned as the likely new limit.) Above 100 MHz, the E- and H-field limits are identical, but there is now a breakpoint in the H-field guidelines, so that the H-field limits are more relaxed at the bottom of the band.

VHF Frequencies.
Figure C shows IEEE exposure guidelines for the frequency range encompassing the VHF television channels. As with the FM-band frequencies, which are in between the low- and high-band VHF channels, the high-band VHF limits are equal for E- and H-fields at 1 and 0.2 mW/cm^2 for occupational and public exposures, respectively. The relaxation of the H-field guidelines for the low-band VHF channels is a continuation of the break begun in the middle of the FM band.

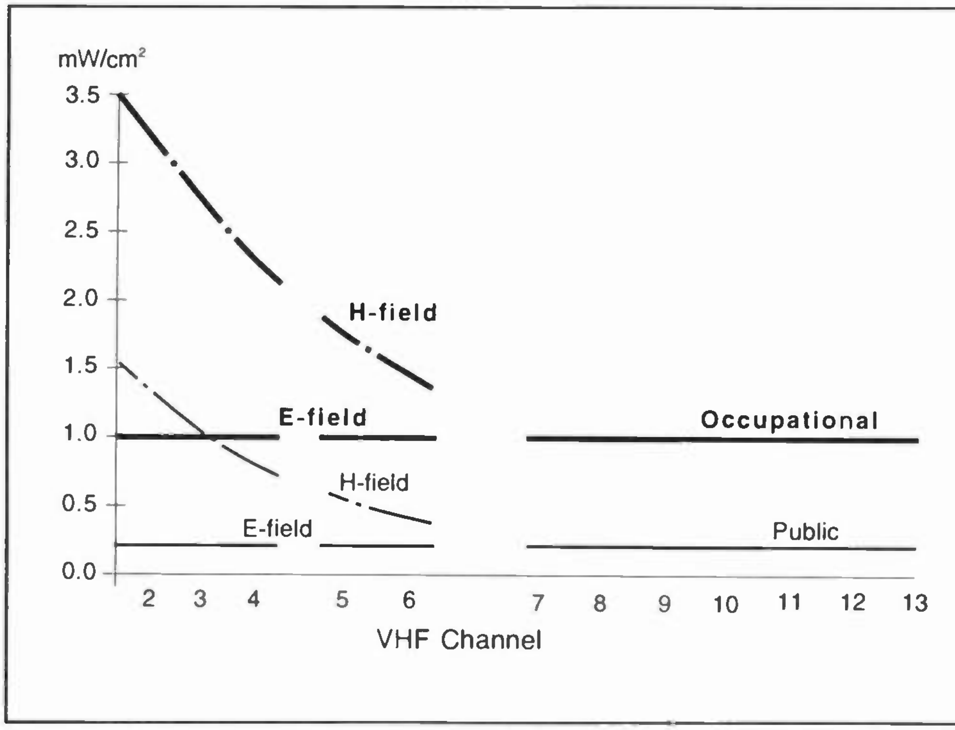

Figure C. IEEE guidelines for VHF TV frequencies.

UHF Frequencies.
The IEEE exposure guidelines are fairly simple for UHF frequencies. The E- and H-field limits are equal, with the occupational limit equal to the ANSI limit and the public limit equal to 1/5 of ANSI. Figure D below shows that both increase smoothly with frequency.

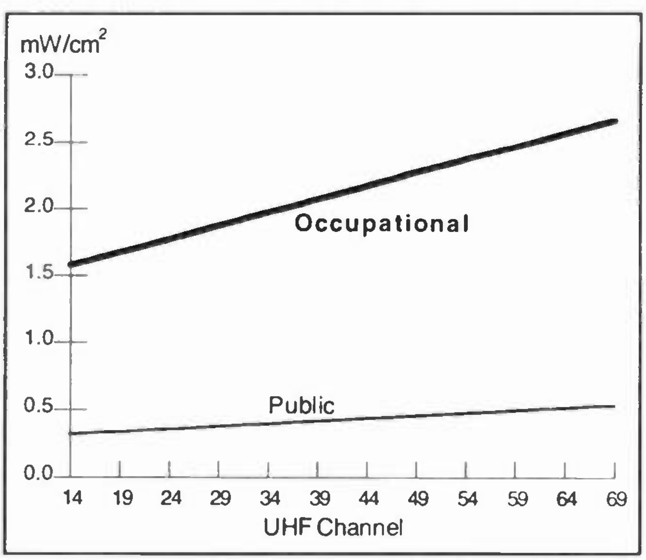

Figure D. IEEE guidelines for UHF TV frequencies.

Microwave Frequencies. The limits for microwave frequencies also do not distinguish between E- and H-fields. As shown in Figure E, the highest occupational limit is 10 mW/cm^2, applying at frequencies above 3 GHz; this represents a two-fold relaxation from ANSI. The public limit does not reach that level until 15 GHz, but public concerns generally are not significant for microwave antennas, because of the small apertures, low powers, and high directivity of most microwave antennas.

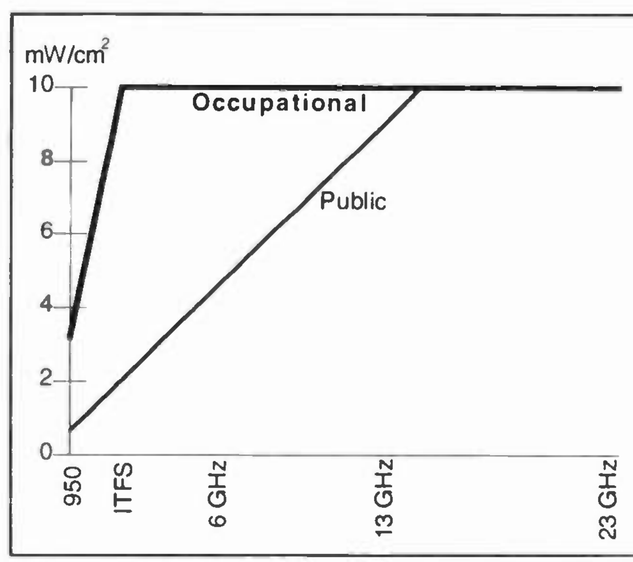

Figure E. IEEE guidelines for microwave frequencies.

BODY CURRENT GUIDELINES

The new limits on the flow of RF current through the human body are perhaps the most controversial of the changes incorporated in the IEEE Standard. Under the

ANSI Standard, measurement of the ambient and localized *fields* was considered adequate to characterize exposures; under IEEE, this is not the case. New limits on body current, whether "induced" in a freestanding individual from the RF environment or generated by "contact" of the individual with a conducting object, require that additional measurements be made at each site.

The occupational limit on either induced or contact current is 100 mA for the AM band, for the low-band VHF channels, and for those FM stations below 100 MHz; there is no limit for stations operating above 100 MHz. Therefore, some FM stations, all high-band VHF stations, and all UHF stations are not subject to a conducted current limit. The public exposure guideline is established with an additional power safety factor of 5, for a limit of 45 mA in the applicable frequency bands.

The main difficulty associated with the specified conducted current limits is determining how to measure the current in practical situations. Measurement procedures are discussed in the next section, but one fact deserves mention at this point. Although the current limits are specified unambiguously in the *text* of the IEEE Standard, the *tables* in the Standard specify two different limits, depending upon whether the current is measured through one foot or two feet, as shown in Figure F. At first glance, the relationship seems to make sense: the current measured through one foot should be half the current measured through both feet, and either measurement should be perfectly acceptable for determining compliance with the Standard. But, in practice, this is not the case.

To illustrate the point, consider the situation of a person standing with both feet on a current meter that is reading 180 mA, which would mean that 90 mA should be flowing through each foot. These (occupational) levels would be in compliance with the IEEE Standard, since the current through each foot is less than 100 mA and the current through both feet is less than 200 mA. If the person now raises one leg, the total current flowing to ground will remain about the same, but all 180 mA is now flowing through one foot; this level is *not* in compliance with the Standard! The fields causing the currents have not changed, but now one concludes that the fields do not comply with the limits. Clearly, it would be difficult to determine compliance with this Standard, since different measurement procedures can yield to completely opposite conclusions. A conscientious engineer could not take the easy way out by measuring just the current through two feet, unless he also required all persons visiting the site to walk only by shuffling both feet along the ground. Therefore, it would appear that, to resolve this particular inconsistency in the IEEE Standard, a specific measurement procedure should be defined; unfortunately, no such procedure is included in the Standard.

MEASUREMENT TECHNIQUES

AM Fields. The H-field limitations at AM frequencies have been relaxed so much that, even though existing broadband probes in that frequency range will now be much too sensitive, measuring fields as high as the limit normally should not be necessary. For the E-field measurements, an integrated meter-probe combination will still be needed to avoid direct lead pick-up (perhaps only until instruments with fiber-optic leads are introduced).

FM & TV Fields. The existing broadband thermocouple or loop/dipole probes will continue to be ideal for measurements of VHF, UHF, and microwave fields, although it is noted that high-frequency H-field probes are not available. This is recognized helpfully in the IEEE Standard by the specification of exposure guidelines for frequencies over 300 MHz in terms of power density only, as opposed to distinct E- and H-field limits. For measurements in the VHF and UHF bands at multi-user sites, using the strictest limit for all frequencies will probably suffice, since there is only a minimal relaxation of this limit at lower and higher frequencies.

Part B - Induced and Contact RadioFrequency Currents Controlled Environments

Frequency range	Maximum Current (mA) Through both feet	Through each foot	Contact
0.003 - 0.1 MHz	2000f	1000f	1000f
0.1 - 100 MHz	200	100	100

f = frequency in MHz

Figure F. Part B from IEEE Table 1.

The results would be conservative, but not overly so, and the measurement procedure is simplified considerably.

Hot-Spots. The IEEE Standard specifies a minimum separation distance from metal objects for measuring localized fields that is four times that specified in ANSI (and two times that allowed by the FCC in MM Docket 88-469). The IEEE limit is 20 cm, as measured from the nearest sensing element of the measurement probe. This relaxation will reduce or eliminate the mitigation measures required under ANSI at many sites, especially those with multiple RF sources.

Time Averaging. The IEEE Standard specifies longer periods for time-averaging only for public exposure conditions, where that latitude generally should not be relied upon, anyway, for mitigating high fields accessible to the public.

Spatial Averaging. In potentially far-reaching paragraphs, the IEEE Standard further relaxes the need to worry about hot-spots: the use of spatial averaging is specified, over an area "equivalent to the vertical cross-section of the human body." Even allowing for worst-case interpretations as to how that area may be projected, considerable latitude is now granted to minimize the effects of localized high fields. The principal restriction is a separate table of "Relaxations for Partial Body Exposures" that allows exposures to localized fields up to 20 times those allowed for the spatially-averaged field.

It is cautioned, though, that this provision applies to "all parts of the body except the eyes and the testes" (which are considered less able to regulate their internal temperatures). Therefore, one should be careful that application of this relaxation of the IEEE Standard will not endanger those body parts in any possible size of person.

Body Currents. Although the IEEE Standard speaks hopefully of "commercial instruments ... beginning to become available," and despite specification of the three allowed methods for measuring induced currents, there will be significant problems with accuracy and repeatability of these measurements (which one would normally expect to be hallmarks of an enforceable standard). Principally, since the human body or its equivalent is itself part of the measurement apparatus, the failure of IEEE to specify standard body characteristics virtually ensures that differently-sized persons will observe (*i.e.*, generate) different readings.

The "bathroom scale" style of meter, recently introduced commercially, is ideal for measuring exposure conditions in situations where an operator would be standing at a single location near an RF-generating device, such as a sealer or welder. The meter would be difficult to use for broadcast site surveys, though, because many readings must be taken around the area to ensure compliance of the entire site. Picking up the meter, moving it, setting it down, standing on it, taking a measurement, and then repeating the procedure numerous times may be good exercise, but it is not tremendously efficient.

In an attempt to simplify these measurements, the author's firm has developed a "boot" style of meter for research purposes, as shown in Figure G. The advantage of this meter is that it can be worn like a boot, allowing an engineer to walk around a site and take measurements at the same time. Meters based on RF current transformers are also under development and are slated for testing within the coming months.

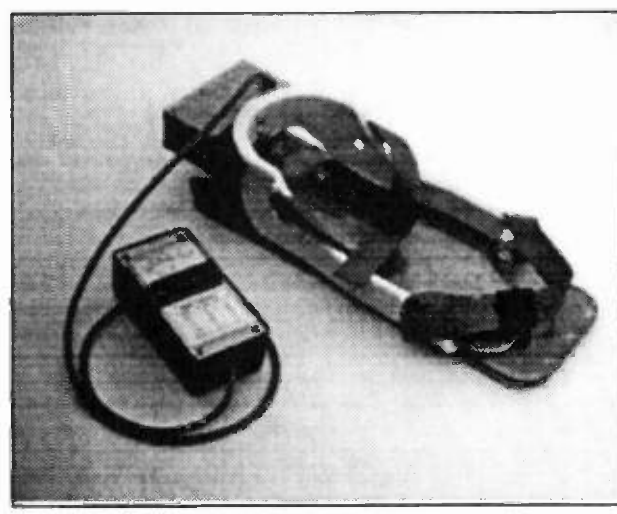

Figure G. RF current-measuring "boot."

The above measurement procedures may appear to some to be overly burdensome, especially since they apply only to stations operating below 100 MHz. It is possible that the frequency range of the body current limits will be scaled back at the time the new standard is adopted by the FCC, or perhaps sooner, to leave out the VHF and FM frequencies. This would have the additional benefit of simplifying the design and construction of accurate measurement equipment, since the devices would have a more limited operating frequency range.

COMPLIANCE TECHNIQUES

Existing techniques for achieving compliance with the IEEE Standard will generally not be affected by the changes from the ANSI Standard. As is the case now, either the field has to be removed from the people or the people from it, or perhaps some combination of the two.

Determination of those conditions requiring such attention is more complicated under the IEEE Standard, because the added body current limits and spatial averaging specifications leave room for individual interpretations of the Standard. Extra care will be necessary to ensure that, if challenged, compliance at a particular site can be demonstrated for all reasonable interpretations of the Standard.

CONCLUSIONS

If the new standard that ANSI intends to adopt is, in turn, adopted by the FCC, additional effort will be required by virtually all broadcast stations to certify that they comply with its new provisions.

Public exposure considerations are exacerbated by the new two-tier nature of the IEEE guidelines and the establishment of five-times tighter public limits. A number of broadcast sites, in anticipation of such stricter standards, have already established public access restrictions at the 200 $\mu W/cm^2$ level. Those that have not done so, though, will have to reexamine, by measurement or by calculation, whether they comply under the new criteria.

Compliance for occupational exposure conditions has often been more difficult to establish but, while additional measurements must be made, the IEEE Standard contains numerous concessions that actually may ease the task of mitigating occupational compliance problems.

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RF PROTECTIVE CLOTHING FOR THE BROADCAST ENVIRONMENT

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This presentation is to discuss a practical approach to electrical field safety from the stand point of a firm whose workers are subjected to electric fields on a daily basis and must protect those workers who are doing the tower climbing, painting and antenna repair.

For many years, some riggers have been willing to assume the burden of responsibility for working in electric fields which were greater than may have been prudent or safe. This option is no longer available to the rigger or rigging company. The FCC has issued guidelines which the company must follow or risk monetary forfeiture. The penalty is not paid by the worker, his employer or the rigging company, but by the licensee. In some case this forfeiture can be substantial. In the case of multiple licensees using the same tower or support structure, the responsibility is shared by all licensees.

The fine can be imposed on all of the licensees, individually or collectively, even if only one is having work done and the others know nothing about it.

The latest standard, IEEE C95.1-1991, is under consideration by ANSI as their standard C95.1-1992. This standard mandates a two tier system whereby one set of standards applies to the broadcaster, his technical employees and contract workers. The other applies to non-technical station personnel and the general public. As far as station technical personnel are concerned, the standard remains essentially the same as C95.1-1982, one milliwatt per centimeter squared ($1 mW/cm^2$) at FM and VHF television frequencies. For non-technical station employees and the general public, the value is reduced to one fifth of this value or 200 microwatts per centimeter squared ($200\ \mu W/cm^2$) in the same range of frequencies.

These two power density values apply to areas called "controlled environment" and "uncontrolled environment". The rationale being that in a controlled environment, technical people should know that there are electric fields near antennas and transmitter facilities. But in the uncontrolled environment other station personnel and the general public may be completely ignorant of the fact that these fields may be present. There are also other substantive changes in the new standard which are beyond the scope of this presentation.

There are several ways to avoid excess exposure to RF fields: Such as controlling the length of time of exposure; removal from the vicinity of the field; shielding and other protective actions; or the use of RF protective clothing such as the Naptex® fabric. Naptex® is the only RF protective fabric currently available for our industry. The FCC is quite explicit in stating that there can be no relaxing of RF compliance rules, intentionally or accidentally. On August 19, 1992 the FCC reissued a Public Notice advisory restating the necessity for compliance.

In the past there have been various attempts to produce protective clothing for workers who are exposed to RF or electric field energy. However, most of the previous versions of protective clothing had one or more drawbacks. For instance the material was too heavy; was made of cloth that did not "breath" (solid material such as aluminum foil) or material that was too cumbersome. Workers would many times refuse to wear the clothing that was provided. Naptex® is the only product to address all these concerns.

As a company which operates with multiple crews working on several different projects, in various parts of the country, at the same time, it is essential that our firm provide its workers with a safe working environment. There are certain inherent dangers in the rigging business which we try to minimize by providing safety belts, other necessary equipment and personal training sessions. It seems only logical then that we also provide RF safety equipment such as protective clothing which will provide a shielding effect from RF fields. Some station owners will provide this equipment and others will require that the contract rigging firm provide the equipment. As with hoists, spray equipment and others specialty items, a lease or rental fee may be imposed by the contracting firm for supplying the RF safety equipment.

It is not enough to insist that prospective tower maintenance contractors supply their own RF suits, the broadcaster must assume an active role by providing a safety program for use by station personnel. Using the station's own suits to assure suit use by the visiting crew can be a part of this program. By maintaining control of the suits to be used, the licensee can have repairs and inspections done in between scheduled projects. If several stations share the tower facility, sufficient number of suits can be available for emergency use. The cost can be shared among all of the broadcasters at the site.

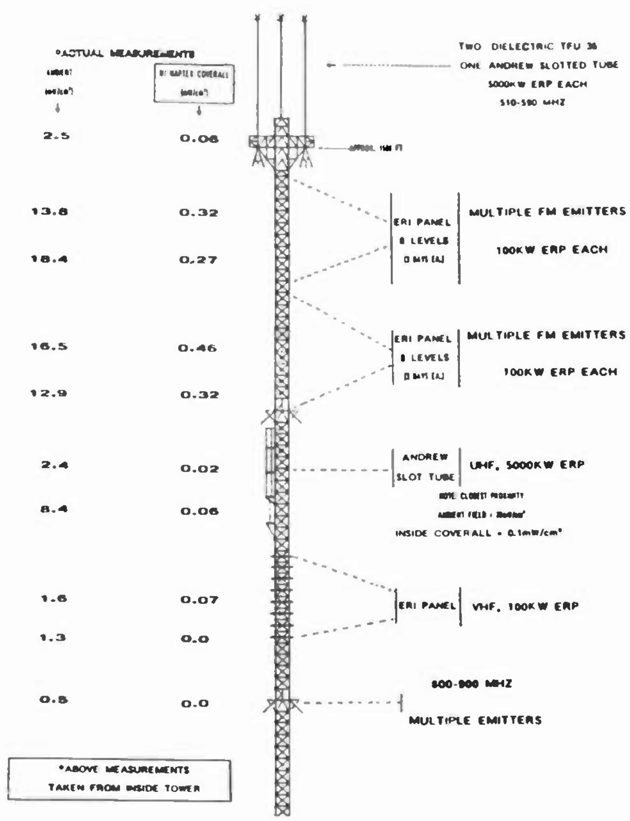
With the environmental movement becoming more of a social force. Various groups have been taking a more negative attitude toward any type of broadcast facility work or tower expansion. A contractor, whose business is servicing the broadcast and telecommunications industry should be well aware of all safety features of tower and facility maintenance.

Remember, it is the broadcaster's responsibility, not the tower crew, for maintaining the safety of the workers. Naturally, this includes RF safety. In addition, the broadcast contractor must be above reproach in the matter of providing a safe worksite for all concerned, including, all equipment which he brings to the job. This means his lifting and hoisting equipment, ropes, loadlines and all safety equipment. Many times there will be an anonymous report to OSHA regarding the work being done. An unexpected visit from an OSHA inspector should be anticipated at any time. If you can show that your personnel are not being subjected to adverse fields and where your fields are higher than specified by the standard that the wearing of a RF suit will reduce the exposure effect to within the prescribed limits, the possibility of a citation can be reduced or eliminated. Documentation of the worksite as to areas over the limit can go a long way in proving that you are making a good faith effort to comply with all the regulations.

RF mapping is a phrase coined to show exactly what areas are off limits to the unprotected worker, or station personnel. This mapping is accomplished by a foot by foot RF measurement of the entire tower to determine unsuitable levels of RF exposure.

Readings would be taken again with the use of the Naptex material to map the areas now accessible if the RF suits are used. Sometimes a tarpaulin made of Naptex® will shield an area where work is being performed. The before and after graphs will form the basis of your RF compliance policy. Record keeping will be a simple matter for the person responsible for tower crew supervision. Because of the bright color used in Naptex® fabric, workers can be seen, from the ground, on even the tallest towers. This can be an aid in safety compliance.

RF MEASUREMENT SURVEY
MULTIPLE EMITTER SCENARIO



MAXWELL SAFETY PRODUCTS,LTD.
20 GILBERT AVE STE 101 HAUPPAUGE,NY 11788

The chart shows a RF map of a typical broadcast tower. You will note that access to the tower is limited to about 900 feet without protective clothing, but that access to the entire tower can be accomplished while wearing the suit. With no change in antennas or related emitting equipment, successive RF maps are not needed. Under most circumstances, only when equipment or power levels are changed would a new RF map be required.

RF mapping as one might imagine, is not something that can be generalized. Each tower installation has a unique set of conditions which require specific measurements and monitoring. Many installations can be mapped in matter of hours and the results are available immediately upon completion. In addition, a RF map of the tower is a good thing to have on hand for inspection by the FCC or OSHA to show your interest in safety compliance. The use of a wide band power density RF measuring instrument does an excellent job of RF mapping.

The possibility of non-compliance by parties not responsible to governing guidelines exposes the broadcaster to an unnecessary risk. Not all contractors have the resources to keep the suits in a condition that guarantees maximum protection, nor are they able to prove effectiveness should the suit become damaged. The suit manufacturer provides regular and emergency service to test and repair the RF suit for a nominal fee. Many tower contractors go from one job to the next with little or no time for minor repairs. A safety inspector will have little sympathy with a contractor who didn't have time to inspect or have his suits repaired. As a matter of policy, the suits should be tested on an annual basis. The turnaround can be within two days.

No one is suggesting that the use of the suit will allow anyone to climb a hot antenna, or allow one to be subjected to really high energy fields directly in an antenna aperture such as a panel array. The manufacturer recommends that it is used in fields no greater than (100 mW/cm²)at 614 V/M. The Naptex cloth can provide attenuation as high as 40 db for frequencies below 1 megaHertz. Attenuation decreases as frequency rises to a value of about 30 db at 1 gigaHertz. Above 1 gigaHertz the attenuation of the fabric falls off to about 10 db at 100 gigaHertz. The manufacturer takes a very conservative position and specifies a minimum of only 15 db attenuation in the FM and TV broadcast frequencies. Field testing has shown that the obtainable attenuation values are actually much higher.

The suits made of the Naptex® material are also very comfortable to wear and there has been virtually no resistance from the tower workers to using the suits. The cut of the suit is such as to allow good arm movement while climbing and the fabric is very breathable. The total effect is about the same as wearing any standard work coverall. Use of the hood has not created too many problems. We have found it best to wear a hard hat or beak cap with the hood which then allows good peripheral vision. There may be some individuals who will complain about the need to wear any protective clothing, but it must be pointed out that there is no alternative. When conditions require the use of protective clothing, it is a essential that it be worn. It must be a condition of employment: wear the safety equipment or look for work elsewhere.

Although there is still some controversy regarding the limits imposed by the electric field standards, even among experts, the standards must be followed or invite inquiry by OSHA, the FCC or local inspectors. As far as electric fields are concerned, we have found the Naptex® material offers the best protection that we know of today. Naptex® was developed in Germany and is imported into this country by Maxwell Safety Products of Hauppauge, New York. The fabric is produced by a double patented process consisting of a stainless steel fiber core surrounded by pure cotton which in turn is surrounded by a polyester/cotton blend. The fabric is remarkably light and does not have the feel of metal fibers as some previous shielding fabrics did. We have found that the material is quite resistant to wear and abrasion and can be laundered repeatedly without any detectable loss of shielding. For more information please call (214) 293-1200.

TAKING MEANINGFUL RFR MEASUREMENTS WITHOUT ENDANGERING THE WORKER

David Baron
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Abstract

Since 1986, ANSI C95.1-1982 has been in effect for broadcasters. With the publication of the revised standard IEEE C95.1-1991 in April of 1992 and its subsequent adoption by ANSI, questions are arising regarding the changes in the standard and their applicability to the broadcast industry. This discussion reviews the changes and the means by which broadcast engineers can safely perform tests to demonstrate compliance with these new requirements.

Introduction

Following adoption of the ANSI C95.1-1982 standard for RFR safety by the FCC in 1986, broadcasters are required to certify that the electromagnetic field levels at transmitter sites do not exceed the maximum allowable exposures. In April of 1992, a revision to the standard was published by the IEEE (Institute of Electrical and Electronics Engineers). This revision (C95.1-1991) was recently adopted as an American National Standard by ANSI (American National Standards Institute). Adoption of the revised standard would require new action by the FCC. While this will likely not take effect in the near future, a lot of interest is developing over the new standard and its possible effect on worker and general public exposure measurements in the broadcast industry.

You can take meaningful measurements of RFR at your facility using your personnel in a manner that will assure compliance with the FCC requirements and safeguard your workers and the general public.

Standards

Two primary changes will affect the way RFR is monitored at broadcast facilities. First the 1991 standard sets two levels of MPE's (maximum permissible exposures) for "Controlled" and "Uncontrolled" environments. These were explicitly not characterized as "Occupational" and "General Public." Rather the Controlled environment is defined as a "location where there is exposure that may be incurred by persons who are aware of the potential for exposure as a concomitant of employment, by other cognizant persons, or as the incidental result of transient passage through areas where analysis shows the exposure levels may be above [the Uncontrolled MPE's] but do not exceed [the Controlled MPE's]."

Second, limits have been applied to RF induced body current due to RFR exposure. These limits apply over the frequency range from 3 kHz to 100 MHz and include the lower portion of the FM broadcast band. The standard applies these MPE's to limit possible shock and burn hazards.

Broadcasters are being visited by OSHA inspectors reviewing workplace safety. The OSHA position is to use the Controlled Environment level as a citation threshold. The Uncontrolled level is viewed as an action threshold. Installations having RFR worker exposures between the two levels are expected to have a formal monitoring program in place to continuously insure that the Uncontrolled levels are not exceeded.

The 1991 standard gives MPE values in field strength units (volts/meter for electric fields [E] and milliamperes/meter for magnetic fields [H]). The 1982 standard stated limits in terms of field strength units squared $|FSU|^2$ (volts^2/meter^2 and amperes^2/meter^2). Both standards also use Plane Wave Equivalent Power Density S [mW/cm^2] as an alternate unit. The following relationships allow conversion between the various units of measure:

$$\frac{E^2[volts/meter]^2}{3770} =$$

$$S[mW/cm^2] =$$

$$H^2[amperes/meter]^2 \times 37.7$$

The IEEE/ANSI Standard is based on a maximum specific absorption rate (SAR) of 4 watts/kilogram. Living tissue can generally absorb and dissipate this level of externally applied energy without a significant increase in temperature. After applying a 10X safety derating factor, field strengths are determined as function of frequency the cause an absorption rate of 0.4 W/kG. These are low levels, well below any threshold of perception. For example, normal microwave ovens will cause SAR's of several hundred watts/kG in water loads placed in the oven.

The acceptable levels for electric field exposure, magnetic field exposure, and induced current exposure are functions of frequency as follows:

Worker exposure can be controlled either by limited RFR levels, limiting exposure times or by controlling access. Reducing RFR levels to the maximum permissible exposures (MPE's) is generally accomplished by reducing or removing transmitter power. In initial characterization testing, it is often desirable to work during the night or early morning hours when transmission can be discontinued with the least disruption.

When limiting exposure times, the broadcaster makes use of the six (6) minute averaging provision of the standard that allows RFR exposures to be average over any six minute period. This section of the standard is most useful when it is necessary for an employee to move through a high field area to reach a protected or low field area where some task must be performed. When tower elevators are used, it is sometimes possible to show that employees, moving through a high field area adjacent to an antenna bay, can safely reach an higher section of the tower where lower field intensities allow extended working times.

Barriers accompanied by warning signs and specific work rules, are an effective means of limiting worker access to possible high field areas. When barriers are used, it is necessary from an OSHA compliance standpoint to clearly mark the area with an RFR caution or warning sign. The warning sign should clearly indicate a standard RFR symbol, state that RF fields exist, and give explicit directions for action. Signs that state "NO ADMITTANCE" can often be interpreted by employees as applying to 'someone else.' An explicit "NO ADMITTANCE WHILE TRANSMITTER IS ON" is much more effective. The third aspect of effective barrier use is a set of written procedures for operating around possible RFR exposure areas. First, they simply tell employees what can and cannot be done safely; giving specific procedures. Second, they present a standard against which you can audit on- going performance. Third, they document your activities for compliance with mandated regulations (FCC and OSHA).

Measurement

Measurement of RFR in relation to human safety is within the expertise of competent broadcast engineers. Equipment is available from several manufacturers that will measure electric and magnetic fields over the full range of commercial broadcast frequencies. These are isotropic devices which respond to the resultant field without regard to polarization. These devices normally readout in units of plane-wave-equivalent power density or field strength units squared $(volts/meter)^2$ or $(amperes/meter)^2$.

Electromagnetic fields at broadcast installations vary with both time (temporal variations) and location (spatial variations). In broadcast locations spatial variations are generally of greater concern than temporal variations. In its simplest form, RFR monitoring consists of sweeping the probe sensor through the space that will/may be occupied by any person. Means such as acoustic headsets provide an indication of relative field strength; allowing the surveyor to find area of high field intensity. Moving the probe from ground level to above head height indicates field maxima due to ground reflection. By extending the probe in the direction of motion, the surveyor can receive prior warning of imminently hazardous fields.

Areas in which fields exceed the MPE's can be surveyed using one of several averaging techniques to determine the average full body exposure level. Localized peak fields (hot spots) must also be evaluated according to the standard. Access to areas where potential exposures exceed the standard must be restricted by operating procedures or physical barriers.

Evaluating exposures on tower structures is done in a similar manner. Using a back mounted field meter with datalogging capability, the surveyor can move about the tower and log actual field intensities. At the same time, it is possible to continuously monitor the moving six-minute average exposure to verify compliance with the time averaged maximum exposure levels. The six-minute average value can be used to predict the point at which the exposure limits may be exceeded.

In the case of a tower elevator, measurements can be made remotely by suitably locating the field measurement instrument and datalogger on the elevator and sending the unmanned elevator through the potentially high field area. On its return, the data can be dumped to a PC or printed out and the field levels determined. It is possible to log and retrieve the six-minute average levels as they vary moving up the tower. In this way it is possible to determine if the time-averaged MPE's will be exceeded by transporting employees up the tower with antennas operating.

The induced and contact current measurements included in the new standard (C95.1-1991) may pose a problem for the broadcaster engineer. The induced current values are a function of individual dimensions and of the effective contact between the individual and "ground." This means taking a range of measurements and carefully documenting your test conditions and results. Two manufacturers currently offer devices for measurement of induced currents. These are "bathroom scale" or mat type devices that are placed between the subject and ground (the person stands on them). Induced current is readout directly in milliamperes. These devices can be used to estimate contact currents if the meter can be located where the subject can stand on it and then take readings under "touch" and "no touch" conditions. Contact Current instruments are under development by RFR equipment manufacturers but are not yet commercially available. The prototype contact current measurement configurations currently being investigated use both phantom impedances to simulate the subject and direct measurement techniques to directly measure contact current flow.

Summary

Effective and safe RFR measurements are possible by competent broadcast engineering personnel. Proper equipment and a basic knowledge of EMF fields is necessary. Careful measurement techniques, meticulous documentation of measurement conditions and test results is essential for results that will stand up under potential outside scrutiny. Documented work procedures and effective signage are necessary to round out an effective RFR control program.

Table 1.

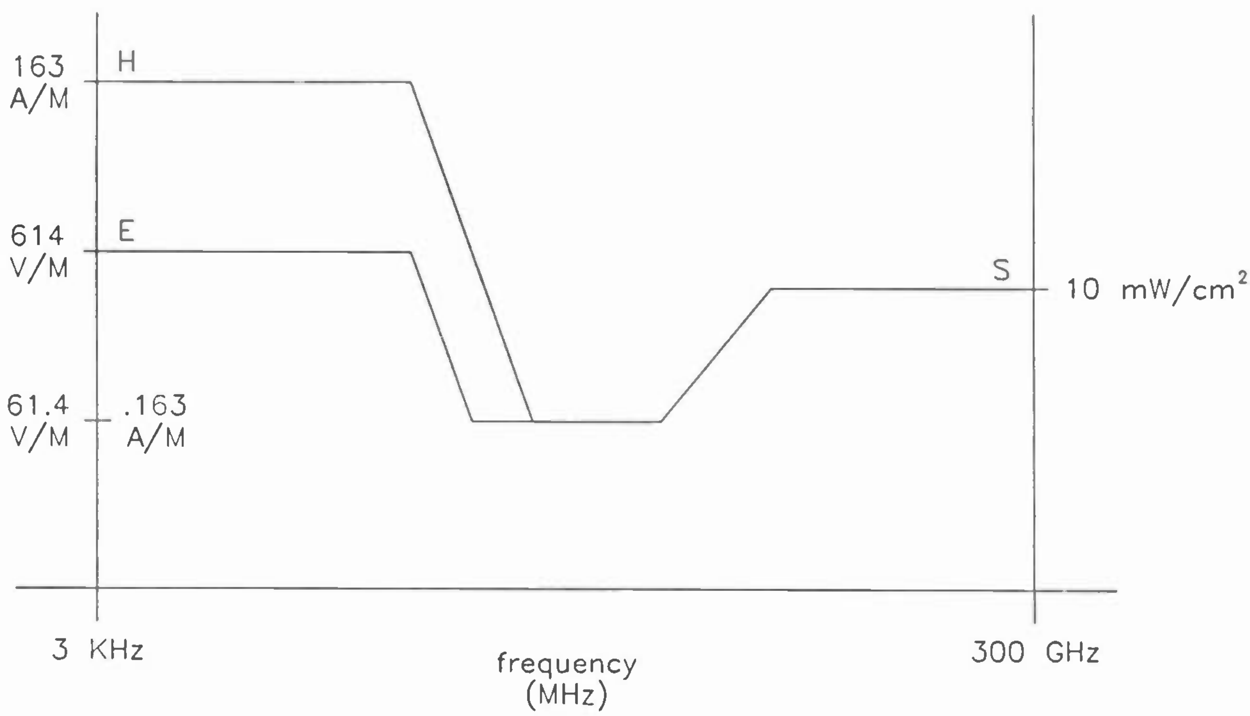
Electric and Magnetic Fields (Controlled Environments)			
Frequency (MHz)	Electric Field (V/M)	Magnetic Field (A/M)	Power Density mW/cm^2
0.1 - 3	614	16.3/f	-
3 - 30	1842/f	16.3/f	-
30 - 100	61.4	16.3/f	-
100 - 300	61.4	0.163	1.0
300 - 3000	-	-	f/300

Table 2.

Induced Current	
Frequency (MHz)	Maximum Current (mA) through both feet
0.1 - 100 MHz	200

Figure 1.

IEEE C95.1-1991
controlled limits



Many municipalities are adopting radiofrequency radiation (RFR) standards that are more stringent than the pertinent federal standards. The FCC will soon decide whether or not to adopt the new IEEE standard C95.1-1991 which reduces the maximum exposure limits to 1/5 of the old limits in "uncontrolled" areas. By using terrain sensitive RFR prediction techniques and by maintaining close cooperation between the station engineer, antenna manufacturer, and engineering counsel, stations can design facilities which not only will meet current standards, but will meet more restrictive standards that municipalities may enact in the future. Working with the municipalities and local citizens to alleviate the problems caused by broadcast facilities is a necessity in the modern business environment.

BACKGROUND

In the Spring of 1984, Ackerley Communications' station in Portland, Oregon, KGON(FM), faced the problem shared by many other stations, how to respond to FCC Docket 80-90 and its subsequent clarifications. Under the new classification scheme, KGON became a Class C1 station. Though KGON operated at 100 kw, its antenna radiation center was 68 feet short of making Class C, and KGON's landlord would not give permission to raise the antenna. Three other tenants on the tower faced the same problem.

The tower on which KGON leased space is located in a tower farm in the Healy Heights area of Portland, Oregon. The seven towers comprising this tower farm are surrounded by an up-scale residential neighborhood, and have been the subject of much controversy. The local citizens have long been bothered by radio frequency interference, worried about radio frequency radiation, and disgruntled by what appears to be uncontrolled growth by an indifferent industry. At the time of Docket 80-90, there were five FM stations located on the same tower as KGON (the "KXL" tower) with radiation centers ranging from 100' to 250' AGL, and one FM station located on an adjacent tower (the "KKSN" tower) with its radiation center at 260' AGL. The KKSN tower was also used as a non-directional radiator for an AM station.

Citizens, fearing the health effects of this congregation of high powered RF emitters, put pressure on city government to investigate the situation. The City of Portland in turn sought assistance from the FCC and the EPA. The outcome of this was a study of the neighborhood undertaken by the Environmental Protection Agency and reported in An Investigation of Radiofrequency Radiation Levels on Healy Heights, Portland, Oregon, July 28 - August 1, 1986, Prepared for the Office of Engineering and Technology, Federal Communications Commission, through Interagency Agreement RW27931344-01-0, by the Electromagnetics Branch, Office of Radiation Programs, U.S. Environmental Protection Agency. Chief Investigator for this study was Richard Tell.

At the conclusion of their study, the EPA found that the highest RF power density in the neighborhood was approximately 700 $\mu w/cm^2$ which occurred near the KXL tower. There were areas within the public right-of-way where the fields exceeded 500 $\mu w/cm^2$. While these levels met the 1000 $\mu w/cm^2$ standard for the frequency band 30-300 MHz as set forth in ANSI C95.1 - 1982, the City of Portland was contemplating adopting a standard equal to 1/5 of the ANSI standard, as had surrounding Multnomah County, and there was strong citizen sentiment to implement a standard equal to 1/10 of the ANSI standard.

Recognizing that KGON's and the other stations' Docket 80-90 problems and the neighbors' concerns over radiofrequency radiation levels could both be resolved by raising the stations' antennas, KGON entered into one and a half years of negotiations with the various neighborhood associations and the City of Portland to find a solution. The outcome of these negotiations was that KGON would construct a new 603' tower, replacing the KKSN tower. KBOO, KWJJ, and KPDQ, the other tenants sharing the KXL tower, would move to the new tower as would KGON and KKSN, and they would share a master antenna system. KXL would remain as the sole occupant of its tower, and the lone AM station was to be moved out of the area. At the heart of the negotiations was the promise that the RFR levels in the Healy Heights neighborhood would be drastically reduced.

The City's proposed regulation procedure for controlling public exposure to radiofrequency radiation was based not on individual emitters, but on power densities at property lines. Since KGON was, in effect, a zoning applicant for five FM stations, and since it shared a property line with the KXL tower, it seemed prudent to design to a power density level that would assure that KGON and its tenants would not contribute more than 50% of the total RFR at any property line. (The EPA study had ascertained that the various land mobile, and microwave users in the tower farm make a negligible contribution to the total radiofrequency radiation field.) Bearing in mind that there was sentiment for a 100 $\mu w/cm^2$ standard, KGON decided to design to a 50 $\mu w/cm^2$ level. KGON specified the combining system and the antenna to handle all of the FM stations then in the area – five full power Class C stations and a 25 kw educational station – and still meet the self-imposed 50 $\mu w/cm^2$ limit.

ANTENNA SELECTION AND PRE-CONSTRUCTION RFR PREDICTIONS

With the assistance of Ben Dawson of Hatfield and Dawson, Consulting Engineers, Seattle, Washington, KGON undertook an investigation into various antennas that could help meet the 50 $\mu w/cm^2$ limit while providing the bandwidth, power handling capability, and performance appropriate to a master antenna system. None of the antennas consisting of stacked, discrete, omnidirectional elements has sufficient bandwidth to handle the frequency spread of the stations involved and meet reasonable VSWR standards. Since KGON was constructing its own tower, it could look at mast mounted designs and not be constrained to considering only side mounted designs, such as panel antennas or cavity backed dipoles.

KGON was particularly intrigued by the JAMPRO JTC spiral antenna. Though this type of antenna had never been used for FM, it has had a long and reliable history in both VHF and UHF TV. The circular symmetry of the design makes for an inherently circular azimuth pattern. Indeed, the manufacturer-supplied azimuth prediction shows a less than ± 1 db variance in both the horizontally and vertically polarized patterns. KGON was particularly impressed with the uniformly low axial ratio in the main beam. The non-resonant, travelling wave design lends itself to the wide bandwidth necessary, and JAMPRO would guarantee the VSWR to be under 1.1:1 over the required 9 MHz bandwidth. Though lower radiation at large depression angles could be achieved with a more traditional panel design with 3/4 λ spacing between elements, KGON felt that the suppression of downward radiation from the JAMPRO JTC was probably adequate and its other attributes more than compensated for its apparently less than optimal performance in this department. KGON decided to concentrate on developing a set of specifications with JAMPRO's assistance for an antenna of this design and then determining from the theoretical elevation pattern whether the ground level radiation limit of 50 $\mu w/cm^2$ could be met. The final design called for 3 spiral elements with 1/2° beam tilt, 20% first null fill and 5% second null fill. This combination produced a gain figure that would allow all of the Class C stations to make 100 kw ERP with no more than 38 kilowatts transmitter power.

KGON developed a chart of power densities to aid in evaluating the antenna design and to allow neighbors to determine what the power density would likely be at their homes from the proposed operation. The power density at any location was determined by finding the intersection of a column specifying the elevation AMSL of the point, and a row specifying the radial distance from the tower. The power densities for each point were calculated using techniques spelled out in the report An Engineering Assessment of the Potential Impact of Federal Radiation Protection Guidelines on the AM, FM, and TV Broadcast Services, by Paul C. Gailey and Richard A. Tell, April 1985, U.S. Environmental Protection Agency, Office of Radiation Programs, Nonionizing Radiation Branch. This technique assumes free space propagation, taking into consideration the elevation pattern of the antenna, and multiplies the resultant power density by 2.56 to compensate for in-phase addition of signals reflected from the ground.

The basic formulas are:

$$P_d = 3.34 \times 10^4 \, ERPD_{total}(\Theta) / R^2$$

$$R = \sqrt{H^2 + D^2}$$

$$\Theta = \arctan(H/D)$$

Where:

P_d = Power density at point of interest in μw/cm^2.

$ERPD_{total}(\Theta)$ = Combined vertically and horizontally polarized power at depression angle Θ referenced to a half wave dipole in kilowatts.

R = Slant range distance between point and radiation center in meters.

H = difference in elevation between the point of interest and radiation center in meters.

D = horizontal distance between point of interest and radiation center in meters.

Using this formula, KGON made a set of predictions using the elevation pattern supplied by JAMPRO at one frequency, specifying a power of 525 kilowatts, equivalent to five 100 kw Class C stations and one 25 kw educational station. Because the supporting structure of the antenna lies entirely within a spiral cage formed by the radiating element, and because the supporting structure is an integral electrical part of the antenna, it was assumed that elevation pattern would not be distorted by the mounting configuration and that the pattern data supplied by the manufacturer was sufficient to characterize this antenna. The highest predicted field was 48 μw/cm^2 occurring at a location just inside the property line, approximately 22 meters NW of the tower. While we felt that the prediction method was conservative, this figure was uncomfortably close to the 50 μw/cm^2 design maximum.

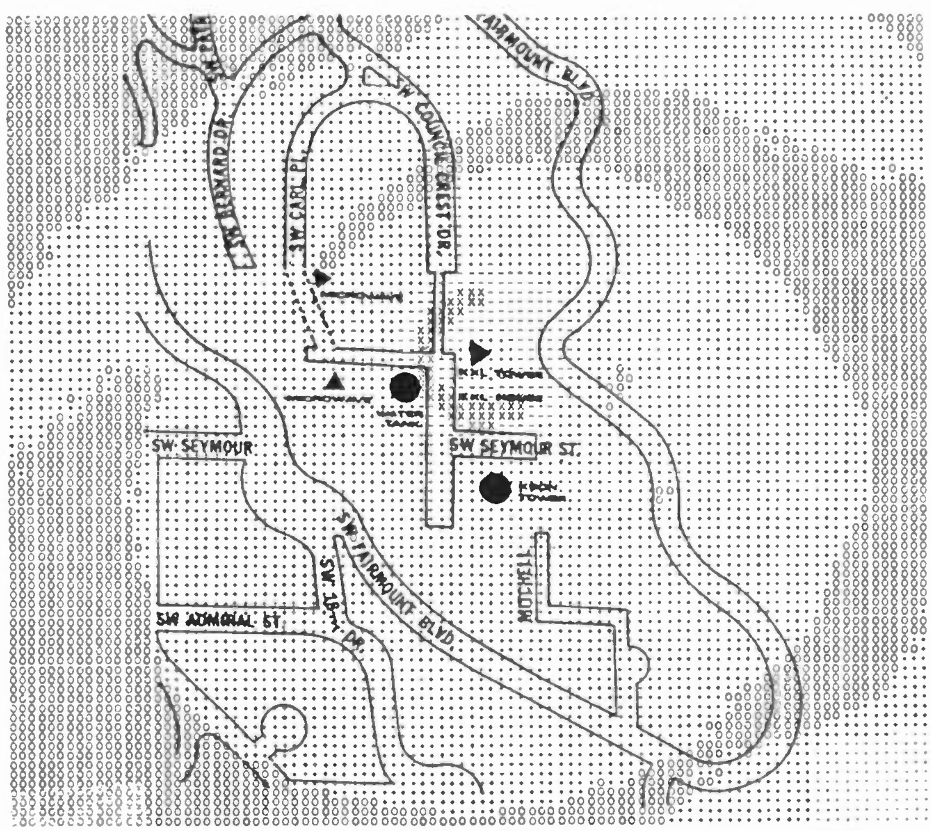
Noting that the maxima and minima of the elevation patterns do not exactly coincide from frequency to frequency, KGON prepared another set of predictions using the elevation patterns of the six individual frequencies with the appropriate powers for each. The highest predicted field using this method is 41.3 μw/cm^2, which occurs 111 meters NW of the tower, indicating that, while KGON would have a little breathing room, the limit was still pretty close.

In retrospect, these charts were not as effective a public relations tool as we would have liked. Most citizens were unable to determine either the radial distance between their home and the tower or the elevation AMSL of their home. This meant that they had to contact the radio station to have us scale this information off a USGS Topographic Quad for them. This was not only cumbersome and disruptive for the station, but prevented a number of people from getting the information they needed. On most of the charts there were points where the power density exceeded the design limit, and, even though these combinations of elevation and distance didn't actually exist, the public had difficulty understanding this, raising unwarranted concerns which were difficult to handle.

Since then, Haertig and Associates has developed a terrain sensitive computer program which calculates the radiofrequency power density at 10,201 points in a 500 meter square grid and then prints a quasi-contour map of the RF density which can be overlaid on a map of the area. This can be handed out to neighbors and they can relate known landmarks and streets to the overlaying power density map. (See Figure 1.) The program uses a terrain data base that is hand digitized from a USGS 7-1/2' Topographic Quadrangle Map at 50 meter intervals and then interpolates a 5 meter interval grid. It was decided that the commercially available 3' terrain databases were digitized at too coarse a level to provide accurate results at the scale needed for mosts RFR investigations.

At the suggestion of Ben Dawson, we decided to undertake field measurements of a similar spiral antenna to verify our prediction technique, the manufacturer's claims, and our assumption that there were no untoward affects on the elevation pattern *in situ*. KTVK – TV, Channel 3, Phoenix, Arizona, employs a 3 element JAMPRO JTC. With gracious assistance from Bill Lawrence, their chief engineer, and introductions and encouragement from JAMPRO, Clint Locey, KGON's assistant chief engineer, and I spent two 100° plus days on South Mountain in April 1989, characterizing the ambient electromagnetic field. South Mountain is a congested electronics site with 4 or 5 television stations and perhaps a dozen FM stations. We made both broadband measurements of the electric field and the magnetic field, and narrowband electric field measurements of both KTVK's visual signal and aural signal.

Measurements were made at series of equally spaced points along roadways or, where possible, overland along straight radii from the KTVK tower. All told, the field was characterized at 82 points. Because of the numerous radiofrequency emitters, the narrowband measurements are of particular interest. These were made using a Potomac Instruments FIM-71 Field Strength Meter and its associated AT-71 dipole antenna. The antenna was adjusted to a compromise length for a frequency half way between KTVK's visual and aural carriers. Error from this is negligible.



Power densities less than or equal to 1.0 µw/cm squared =
Power densities less than or equal to 10.0 µw/cm squared = O
Power densities less than or equal to 50.0 µw/cm squared = +
Power densities less than or equal to 100.0 µw/cm squared = -
Power densities less than or equal to 200.0 µw/cm squared = X
Power densities greater than 200.0 µw/cm squared = •

Greatest predicted power density is 120,4 µw/cm squared at row.41, column 46.

Courtesy Haertig & Associates

Figure 1. Predicted radiofrequency power densities on Healy Heights using theoretical elevation pattern for KXL and original elevation data for JAMPRO spiral antenna.

The antenna was secured to a homemade mount which placed the center of the antenna at 2 meters AGL and positioned it at the so called analytical angle, 54.7° with respect to vertical. This is the angle made between the diagonal and an included side of a cube. Since the three sides of a cube that intersect at its diagonal are at right angles to each other, and since the angles between the sides as projected onto the plane perpendicular to the diagonal are all 120°, by simply rotating a dipole that is aligned with one side of the cube around the diagonal through three positions 120° apart, one can make a set of mutually orthogonal measurements without having to change the antenna mounting. These measurements were recorded and later multiplied by the antenna correction factor and the antenna cable correction factor. The root sum square was then taken of the three measurements which gives the magnitude of the field vector at that point.

The field strength was then converted to power density. Where it can be assumed that the E and H fields, and the direction of propagation are mutually orthogonal (i.e., "far field" conditions exist), the power density at a given point is related to the field strength by the relationship:

$$S = E^2/377$$

Where:

S = the power density in W/m^2 and
E = field strength in V/m.

All the measurement locations were at least 10 wavelengths away from the antenna, so it can be reasonably assumed that far field conditions prevailed. There are some inherent errors in the measurement technique used: the AT-71 antenna is only calibrated in the horizontal position at 9' and 30' above ground; the antenna lead was not kept orthogonal to the antenna, so there is likely some interaction; the antenna size is sufficiently large that the power density is not uniform over its entire length (free space conditions don't exist); and the measurements are taken at only one elevation and therefore cannot characterize either the peak field or a spatial average. The effects of the first three items are likely to be random so that there is a tendency for the errors to cancel in a large number of measurements. Experience taking numerous broadband measurement has shown that the field near 2 meters, absent reflecting objects, is generally not changing very rapidly with distance from the ground and is at some mid-value between the peak and minimum readings. We believed this to be sufficiently accurate to determine whether the prediction method is suitably conservative.

The highest fields measured were 7.7 $\mu w/cm^2$ for the visual carrier and 0.66 $\mu w/cm^2$ for the aural carrier. We made an attempt to back calculate the pattern of the antenna from the measured power density data, but the scatter was large and the results weren't quantitatively meaningful. However, the data taken at the visual carrier frequency show a general increase in the field in areas that correspond to antenna depression angles near 60° and 40°. This corresponds nicely to the position of the side lobes for this antenna as measured on the manufacturer's test range. There were corresponding lobes in the pattern at the aural frequency at depression angles near 33° and 55°, thus corroborating that the shape of the elevation pattern changes with frequency.

Because the scatter in the data made it difficult to do a detailed analysis of the antenna response, we decided that a worst case analysis might be sufficient to allay our fears. The approach was to take the highest measured aural and visual power densities and scale them to the proposed facilities on Healy Heights. As discussed above, power density is directly proportional to power transmitted and inversely proportional to the square of the distance to the radiation center. The scaling factors are calculated by:

$$\text{Power} = \frac{ERP_{\text{Healy Heights}}}{ERP_{KTVK}}$$

$$\text{Height} = \left[\frac{H_{KTVK}}{H_{\text{Healy Heights}}}\right]^2$$

While this technique is crude at best, it seems a reasonable first order approximation. With a KTVK aural carrier power of 15 kw and a proposed Healy Heights total power of 525 kw the power scaling factor equals 35. For the visual carrier power of 100 kw, the scaling factor becomes 5.75. The height scaling factor is .2592, with KTVK's radiation center at 280' AGL and the proposed Healy Heights radiation center at 550' AGL. Scaling the maximum KTVK aural carrier measurement, 0.66 $\mu w/cm^2$, to Healy Heights gives 5.99 $\mu w/cm^2$, and scaling the maximum visual carrier measurement, 7.7 $\mu w/cm^2$, gives 10.5 $\mu w/cm^2$. The scaled aural carrier level is 9.2 db down from the design limit of 50 $\mu w/cm^2$ and the scaled visual carrier level is down 6.8 db. This is considerably more leeway than the 8.7 $\mu w/cm^2$ (0.83 db) indicated by the prediction based on the theoretical elevation pattern, and KGON felt sure that, as far as downward radiation was concerned, the JAMPRO antenna would perform better than the manufacturer's predictions would indicate and that our method of predicting the ground level radiofrequency power density on Healy Heights was sufficiently conservative to assure that we would meet the self-imposed limit of 50 $\mu w/cm^2$.

JAMPRO has since revised their computer model of the spiral antenna, taking into consideration the tapered illumination of the individual spiral elements. I made another set of predictions using the new elevation patterns supplied by the manufacturer. With a total of 525 kw ERP, the highest predicted ground level power density is 9.6 $\mu w/cm^2$. I suspect that, had these data been available at the time, KGON would still have undertaken the South Mountain study, attributing the elevation patterns to the manufacturer's rosy spectacles. Note, however, the scaled values from the South Mountain agree within 1 db with the maximum value predicted using the new data.

PRE-CONSTRUCTION RFR MEASUREMENTS

Prior to and during the erection of the KGON tower, the local neighborhood associations were lobbying the city for a long range plan for the Healy Heights tower farm in order to address their concerns about the future. Central to their concerns was establishing a method of on-going radiofrequency radiation monitoring. Amongst the members of the neighborhood associations are several physicians and research scientists, who, because of their training and scientific bent, felt that it was of prime importance in any monitoring scheme to establish a baseline of data to compare future measurements to, and to establish a repeatable protocol to assure that future measurements were comparable. While, from a regulatory standpoint, this was unnecessary – after all, all that one is concerned with is whether a facility, as built, will subject people to radiofrequency fields in excess of the standard – there is no arguing that the data would be interesting to have. Several members of the broadcast industry balked at the idea of paying for such a study, and the city indicated that it was not in a position to finance it. After some cajoling by one of the City Commissioners, a compromise was struck wherein each broadcaster, each non-broadcast tower owner, and the city would pungle up equal shares to fund the study. (As its share, the educational station contributed the services of this office (provided to them on a *pro bono* basis) and of several volunteers, to assist in the measurement process.)

The City solicited proposals from several firms to create a measurement procedure and to perform the measurements. The citizens groups expressed reservations about using engineering firms connected with the broadcast industry, fearing that they might not be sufficiently disinterested, and lobbied heavily for hiring Richard Tell Associates, Inc., of Las Vegas, Nevada. They were aware of Mr. Tell's national reputation in the investigation and measurement of electromagnetic fields, and were impressed by his candor and competence when he led the team making radiofrequency radiation measurements in Healy Heights for the Environmental Protection Agency. His familiarity with the area and prior history making measurements there further recommended him to the City.

Ultimately the City hired Mr. Tell to create a repeatable protocol for assessing the electromagnetic environment in the Healy Heights area and to make a two phase study of the power densities found there. The phase one measurements were to be made prior to the activation of the stations that were to move to the new KGON tower, and the phase two measurements were to be taken after all of the stations became operational.

Repeatability was of prime concern in designing the measurement protocol. To that end, Mr. Tell decided that the measurements in the first and subsequent phases should be taken at the same points and that the points should be selected non-subjectively without reference to the expected power density; the number of points measured should be sufficiently large to be able to statistically represent the power levels in the entire neighborhood; and the points should be marked and located in such a fashion as to assure their relocation for future measurements. The specific method of handling the measurement instruments should be spelled out so that it can be repeated by other operators and the actual data acquisition should be largely automated to remove subjective interpretation by the operator.

The measurements were divided into broadband and narrowband measurements. Since the primary concern of the residents and the City was the aggregate radiofrequency power density, the bulk of the measurements were broadband in nature. characterizing the electromagnetic field without regard to individual emitters. The broadcasters were particularly interested in the relative contributions of the several emitters, so at a subset of the broadband measurement points, narrowband, frequency specific measurements were made. Mr. Tell knew from his earlier measurements taken with the EPA that the contribution to the aggregate power density of the numerous land mobile and operational fixed microwave operators in the neighborhood was very small. This study included a less stringent examination of these fields to verify his previous experience. It was found that the worst case contribution from these sources might be 5.3 $\mu w/cm^2$ with a more likely value to be about 0.75 $\mu w/cm^2$. From this it can be concluded that, as far as compliance with radiofrequency radiation exposure guidelines is concerned, these signals are unimportant on Healy Heights and further study of them is probably not warranted.

The first phase measurements were taken over a two day period in July 1990. During the first day, Mr. Tell and I laid out the location of the points to be measured. As far as possible the points were located in the center of public roadways to assure continued access and to avoid the field perturbing influence of nearby objects. In the immediate vicinity (approximately 500 meters) of the broadcast towers, measurement points were established every 20 feet. At further distance, the point spacing was dropped to every 50 feet. There were also a handful of points taken along property lines and other select areas chosen primarily for their interest to the neighborhood (such as the local playground) or to fill in areas where there weren't public roads. Each point was marked by a cross of surveyor's orange paint. One arm of the cross was located parallel to the roadway and one perpendicular. Every fifth point was numbered with the same orange paint. A second crew came along later and drove a surveyor's nail into the asphalt at the center of each cross to ensure that the points could be relocated even after the paint had worn off. All told, 171 measurement points were established.

At each of these points the electric component of the aggregate ambient electromagnetic field was measured with a Holaday Industries, Inc. HI-3001 Radiation Hazard Meter with the companion GRE electric field probe. This instrument was chosen because of its long and short term calibration stability, its rapid response to changes in the electric field, and the very wide dynamic measurement range available at its external recorder output without range switching. The probe consists of three mutually orthogonal dipoles with diode detectors which are summed in the probe. It has a specified isotropicity of <.5 db and the calibration accuracy is .5 db. Unfortunately, the diode detectors do not display a true RMS response and the instrument will generally read high in the presence of multiple radiofrequency fields. The readings on this instrument were compared to the readings obtained at several representative points using a NARDA Microwave Corporation model 8700 Radiofrequency Radiation Test Set, which is known to have a nearly RMS response. The Holaday meter read from 15% to 73% higher than the NARDA meter. This is consistent with a conservative approach to the study. The NARDA meter was not chosen for making the broadband measurements because of its slow response time (up to 3 seconds) and because field calibrations must be performed frequently and require an enclosure for the probe to assure that it's effectively in a zero field.

The power density varies considerably with distance from the ground because of additive and subtractive reinforcement of the incident field with the field reflected from the ground. There can be problems with repeatability between measurements by different observers if there is not agreement as to what point in space is actually being measured. From a regulatory standpoint, there has been considerable discussion as to how to specify the magnitude of the field when it is spatially variable. The recently adopted IEEE C95.1-1991 standard looks at exposure standards in terms of a whole body spatial average of measured power density. In practice this means making several equally spaced measurements in the volume that would be occupied by a human body, and then taking an algebraic average of them. To deal with this problem, Mr. Tell's protocol called for the instrument's external recorder output to be connected to a data logger that was capable of recording four measurements per second. The logger was set to record data over a ten second interval as the probe was swept vertically at a constant velocity from ground level to a height of 2 meters, thus producing forty equally spaced data points per sweep. The data logger then calculated the average of the forty points and recorded the average as well as the maximum and minimum measurements. The ambient fields are perturbed by the presence of the instrument operator and the instrument itself. In order to average out the effect of the operator's presence, four sweeps were performed at each point with the operator moving around the point 90° between each sweep. In general, the operator lined himself with each of the four arms of the cross marking each point with the sweep made directly above the intersection. Periodically the data were downloaded to floppy disk for later analysis using a laptop computer connected to the data logger's RS-323 serial port.

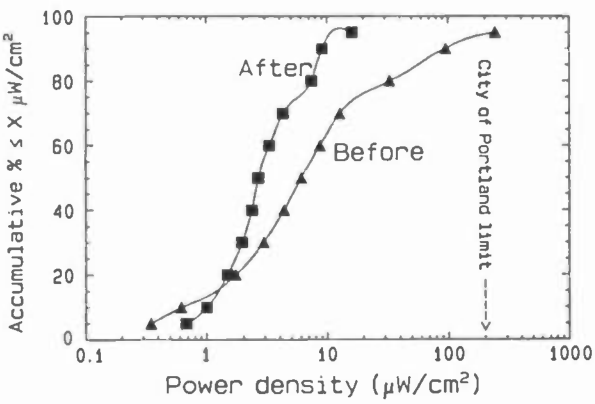
Narrowband measurements were performed at 10 points, 9 of which were a subset of the broadband points. These measurements were made using the same technique used by Clint Locey and myself for the narrowband measurements made on South Mountain in Phoenix, as detailed above. Measurements were made at each of the six frequencies used by the FM broadcasters on Healy Heights. The FIM-71 Field Strength Meter has a rated accuracy of ± 3 db. After each narrowband measurement, the dipole was removed from its stand and broadband measurements made with both the Holaday and NARDA meters with their respective probes placed at the location where the center of the dipole antenna had been.

The NARDA meter was then used to map the extent of the 200 μw/cm2 power density contour around the KXL tower. This corresponds to the City of Portland RFR limit at FM frequencies. Measurements were taken in 3 directions corresponding to the location of public streets relative to a survey marker in the middle of the street due west of the KXL tower; the forth direction was through a briar patch. Finally the NARDA meter was used to find the point of highest power density regardless of location.

During the analysis of the broadband data, the minimum, average, and maximum readings from each of the four sweeps per point were averaged to provide single values for these parameters for each point. The points were then sorted into order by power density, and the data presented as the percentage of points whose power density was less than a given value. 50% of the points had spatially *averaged* power densities less than 6.05 μw/cm^2 (the median value); 90% were below 95.5 μw/cm^2; 95% were below 242 μw/cm^2; and the highest spatially averaged value at any of the 171 points was 488 μw/cm^2. (See Figure 2.) 50% of the points had a spatial *maximum* value less than 10.2 μw/cm^2; 90% were below 143μw/cm^2; 95% below 464 μw/cm^2; and the highest maximum value at any of the 171 points was 897 μw/cm^2. The highest broadband power density that was found anywhere was between 1385 and 1556 μw/cm^2, the field being unstable enough that it was difficult to get an accurate reading. This is not a spatially averaged value. The 200 μw/cm^2 contour was found to extend well into public areas thus putting the KXL tower property owner in violation of City ordinance.

At each frequency at each data point used in the narrowband measurement the RSS (root-sum-square) value of the field vector was computed from the three orthogonal measurements. These field strengths were then converted to power density values and summed to provide the aggregate power density at the point. These aggregate power densities ranged from 1.84 μw/cm^2 to 206 μw/cm^2. The individual power densities were also expressed as a percentage of the aggregate. The broadband measurements taken using the NARDA meter with the probe occupying the same position as the center of the dipole used in the narrowband measurements ranged from -2.6 db to +1.5 db relative to the summed narrowband measurements, with the majority of the measurements within 1 db. This is well within the calibration accuracy of the instruments. Surprisingly, the measurements made with the Holaday meter also showed a good correlation with the range running from -3.4 db to +2.7 db. In 9 out 10 measurements, KWJJ contributed the majority of radiofrequency radiation, with their share ranging from 26.7% to 55%. This was consistent with their occupying the lowest position on the tower. There was considerably greater variation in the contributions of the other stations with percentages ranging from .7% to 50.3%. The radiation centers of the other five stations were all considerably higher than KWJJ's so that variations in the array patterns dominated the variation in power density at a given location. Since none of the radiation centers were collocated, the nulls and lobes fell in different places at ground level.

Mr. Tell delivered his report, An Investigation of Radiofrequency Fields on Healy Heights, Portland, Oregon, Phase I, to the City of Portland on 14 August 1990.



Courtesy City of Portland & Richard Tell Associates, Inc.

Figure 2. Comparison of broadband spatial average power density between Phase I and Phase II in terms of accumulative percent of measurement points exhibiting power densities < value on horizontal axis.

POST-CONSTRUCTION RFR MEASUREMENTS

Mr. Tell returned to Portland in November 1991 to complete the second phase of his study. This was shortly after the fifth and final station had commenced broadcasting from the new tower. Prior to his arrival, I went out with broom and surveyor's paint to rediscover as many of the original points as possible. I was able to find all but ten of the points. Mr. Tell and I then approximately located those points using measurements off of known points and memory. Because of the statistical nature of the study, the location errors of these few points does not seriously compromise the results of the study.

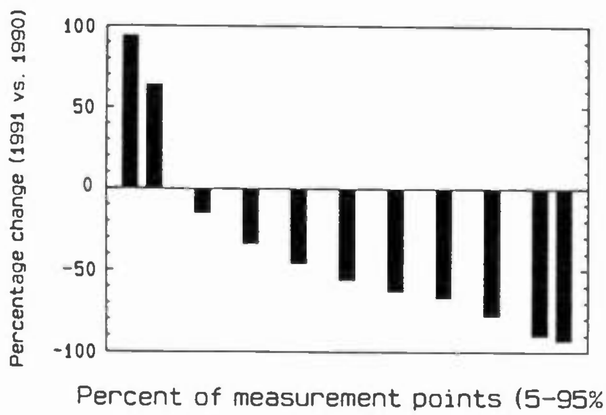
In a one day period between cloudbursts, we were able to repeat broadband measurements at all 171 points, repeat the narrowband measurements at the ten points chosen in phase I, and locate the highest field in the area. The data was analyzed in the same fashion as in phase I and then compared with the results of the first phase.

The median value of the spatially *averaged* power density in the second phase of the study is 2.66 $\mu w/cm^2$; 90% of the points fall below 9.08 $\mu w/cm^2$; 95% fall below 16.1 $\mu w/cm^2$; and the maximum average power density at any of the 171 points is 36.02 $\mu w/cm^2$. (See Figure 2.) The corresponding values of the spatial *maxima* are: 50% below 4.54 $\mu w/cm^2$; 90% below 14.3 $\mu w/cm^2$; 95% below 25.2 $\mu w/cm^2$ with the maximum field recorded at any of the 171 points being 69.61 $\mu w/cm^2$. The absolute highest power density level that could be found in the phase II study is 98 $\mu w/cm^2$, 12 db below the corresponding field found in phase I.

There was a reduction of 7 to 8 db in the aggregate power densities computed from the narrowband measurements between phase I and phase II with the power densities ranging from .26$\mu w/cm^2$ to 39.2$\mu w/cm^2$. As might be expected because of its now relatively low height, KXL, the sole remaining station on the KXL tower, contributes the majority of the power at eight out of ten measurement points with percentages ranging from 5.3% to 97.1%. The 97.1% share occurs at the narrowband point that has the highest aggregate power density.

Comparing the phase I and phase II broadband measurements shows a pronounced shift of the overall power density in the study area towards values in the 1 to 10 $\mu w/cm^2$ range. (See Figures 3 & 4.) In phase I 52.6% of the points fell within this range while in phase II 82.5% fall there. In phase I 17% of the points fell within the 50$\mu w/cm^2$ to 500$\mu w/cm^2$ range, and in phase II there are no points at all in this range. Interestingly, the number of points with power densities less than 1$\mu w/cm^2$ has actually increased. This reflects the fact that many of the more distant points that were formerly behind local terrain obstacles now enjoy line of sight to the taller, new tower.

Mr. Tell delivered his second phase report to the City in December 1991 and formally presented his findings to the City Council, the engineering representatives of the tower owners and broadcasters, and the neighborhood association in a series of meetings in January 1992.



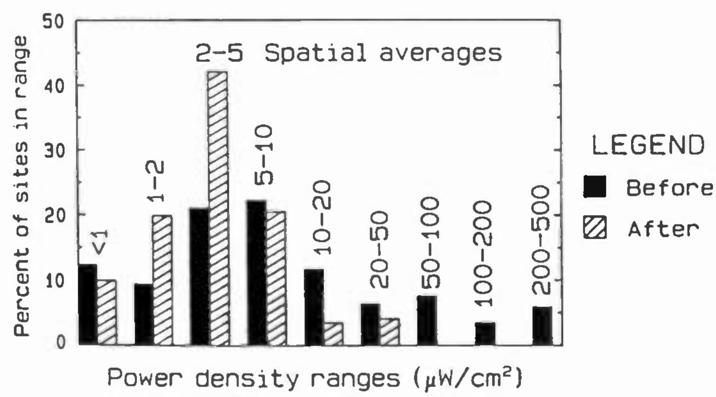
Courtesy City of Portland & Richard Tell Associates, Inc.

Figure 3. Analysis of changes in measured spatial average power densities between phase I and phase II in 5, 10, 20, 30, 40, 50, 60, 70, 80, 90, and 95 percentile ranges.

COMPARISON WITH PREDICTIONS

In a general sense, the broadband measurements taken in phase II agree with the predictions made using the revised elevation patterns from JAMPRO. The prediction shows that 85% of the 10,201 prediction points should fall within the 1 $\mu w/cm^2$ to 10 $\mu w/cm^2$ bracket and in the phase II measurements, 82% of the 171 measured points fell within this same bracket. According to the prediction, the power density at 13.3% of the data points should fall between 10 $\mu w/cm^2$ and 200 $\mu w/cm^2$, and the phase II measurements revealed that 7.6% fell within this category. The maximum predicted power density level, 108.8 $\mu w/cm^2$, is only 0.45 db higher than the measured maximum value of 98 $\mu w/cm^2$.

The predictions made using the original elevation pattern data supplied by JAMPRO tend to overpredict the power density level. The prediction suggested that 40% of the points should fall in the 1 $\mu w/cm^2$ to 10 $\mu w/cm^2$ bracket, 57% should fall in the 10 $\mu w/cm^2$ to 50% $\mu w/cm^2$ bracket, and the maximum power density level would be 120.4 $\mu w/cm^2$. The difference between the predictions using the original data and the revised data is not as large as might be suggested by the sets of predictions made for just the KGON tower alone. This is because the contribution from KXL tends to dominate the power density at most of the points.



Courtesy City of Portland & Richard Tell Associates, Inc.

Figure 4. Comparison of percentages of measurement points with power densities in various ranges based on spatial average values in phase I and phase II.

Seven of the ten narrowband measurement points are within the geographical bounds of the prediction studies and are also a subset of the broadband data points. In general, the spatial maximum of the power density for each of the points as measured in the broadband study agrees nicely with the summation of the individual power densities of the narrowband studies. The broadband values ranged from -0.69 db to +6.3 db relative to the narrowband measurements for all seven points and -0.69 db to +1.8 db for six points.

As part of the research for this paper, I made a prediction of KXL's contribution to the total power density area using a cosine vertical element pattern and a theoretical five element array pattern with wavelength spacing and equal power distribution and phasing. At five out of seven of the points, this model significantly overpredicted the power density, with the levels of the prediction ranging from -12 db to +17 db relative to the measured value over six points. The seventh point has a predicted power density of zero. Over five points the range is +1.2 db to +17 db. Both of the points where the predicted value is less than the measured value have very low predicted values even though the points are relatively close to the KXL tower. The low values are the result of a null in the theoretical array pattern. In practice, reradiation from the antenna support structure and the multiplicity of objects in the antenna's field tends to fill these nulls and the ground level power density, while generally reduced relative to the adjacent lobes, has an appreciable value. Prudence dictates that the array pattern used for these predictions should be an envelope of the theoretical pattern. Alternatively, one must interpret the prediction in light of this defect.

Looking at the contribution of only those stations using the JAMPRO antenna, at seven out of seven points the predictions made using the *original* elevation data overpredicted the measured amounts by significant factors with the predictions exceeding the measured by +6.9 db to 20 db. At six out of seven points the predictions using the *revised* patterns exceeded the measured over a range of -0.67 db to +16.9 db with the five point range being -0.67 db to +4 db.

CONCLUSIONS

The use of a terrain sensitive ground level radiation prediction technique proved invaluable in meeting KGON's design goals of limiting its contribution to the overall radiofrequency radiation level in Healy Heights. However, because of the paucity of data about the performance of normal mode helical antennas *in situ* and the lack of a good theoretical model to describe them, field verification of the elevation pattern data was essential in determining the validity of the process.

Gailey and Tell, *supra*, discuss in detail the effects of the antenna mounting structure on both the elevation and azimuth patterns of FM broadcast antennas. Their study focuses primarily on the effects of the mounting structure on the individual element patterns. The array pattern is also distorted by the mounting structure and there is no good data on these affects. Though difficult, it is possible to calibrate an antenna test range to sufficient accuracy to obtain a good representation of an antenna array's azimuth pattern at FM frequencies. However, accurate elevation pattern data of whole antenna arrays as mounted are nearly impossible to obtain at these frequencies. Because the antenna sweeps a much larger volume when rotated in the elevation plane than the azimuth plane, the test range must be able to produce a much larger volume of uniform field in order to produce accurate results. Because the linear dimension of this volume is several wavelengths, and because the wavelength at 100 MHz is near 10 feet, ground reflections destroy the uniformity of the field for any economically realizable test arrangement. Bob Surette of Shively Laboratories, indicates that, while it is possible to make scale model array measurements of FM antennas at 4.5:1 (approximately 450 MHz), it would be an undertaking to build a test range of sufficient size and uniformity to do so. Shively Labs is, to my knowledge, the only FM antenna manufacturer regularly making scale model measurements of FM antennas, and, according to Mr. Surette, their range can only guarantee a uniform field in an aproximate 1 meter cube, sufficient for measuring the elevation pattern of but one element.

This reinforces the necessity of obtaining a body of field data for various FM antenna arrays in various mounting situations. I would like to see the industry undertake a study similar to the Gailey and Tell study, to document the element patterns *in situ* of available antennas and the effect of mounting structure on the *array* pattern. Absent this data, the design engineer must exercise a degree of caution in using theoretical predictions as a design tool in meeting whatever radiation standards are in force in the area of operation. As seen in the example of KXL, above, the designer cannot rely on the depth or accurate location of elevation pattern nulls in order to meet the standards, and should use an envelope of the array patterns. Pattern distortions are particularly severe for arrays of individual omnidirectional elements side mounted on towers. An examination of several azimuth plots for various mounting configurations shows that there can be as much as a 4 db increase in the field over a true omnidirectional pattern. Prudence would dictate that a safety margin of at least 10 db should be left between the highest predicted field and the design standard.

Not only do the post construction RFR measurements confirm the predicted reduction in RFR, but they now form a baseline on which to judge the suitability of future development in the Healy Heights tower farm. The City of Portland has recently adopted a long range plan for this area, and continued RF monitoring and prediction is fundamental to it. As future broadcasters or other high power RF emitters apply for the conditional use permit required to construct transmission facilities in the area, they will have to supply predictions of the ground level radiation caused by their project. If the predicted peak ground level power density is less than 1/4 of the difference between the last set of measured data and the applicable standard, then it is assumed that the project as built will meet the standards. However, if it is greater than 1/4, a new round of post-construction measurements is triggered at the applicant's expense. This then becomes the new baseline for future decisions. This scheme assures that monitoring occurs more frequently as the radiofrequency radiation limit is approached, but does not subject the industry to a lot of unnecessary monitoring.

Many municipalities are adopting radiofrequency radiation guidelines more stringent than the federal standards, and in many areas, urban growth has put broadcast facilities cheek by jowl with residential areas. The public is now much more concerned with the health, interference, nuisance, and aesthetic impact of these facilities than they have been in the past. The broadcast industry, like many other industries, has enjoyed a long period where it has not had to be responsible for the detrimental effects it has on the public. This era is ending, and the more that broadcasters resist these regulatory changes, the less they will be able to contribute to them. KGON has found that in dealing honestly with the public, in not belittling their concerns, and in going beyond the minimum required to appease them, KGON is treated fairly as a neighbor and its opinions are sought out and frequently believed in developing the regulatory process. Where in the past each change on Healy Heights has been met with strong opposition, involving considerable bureaucratic wrangling and legal cost, now a plan is in place that assures the continued health and growth of the communications industry while protecting the health, liveability, and peace of the neighborhood.

Through careful engineering, close consultation with the manufacturer, and a corporate commitment to being responsible for the pollution generated by its business, KGON has been able to not only reduce the peak ground level radiofrequency radiation from 1556 $\mu w/cm^2$ to 98 $\mu w/cm^2$ (12 db), but to make great strides towards turning around twenty years of neighborhood distrust and anger.

ACKNOWLEDGEMENTS

The author wishes to gratefully acknowledge the assistance of the following people in filling the vast lacunae in his knowledge and for letting him ride their coat tails to glory. All errors of omission, commission, or transmission are, of course, not theirs, but the computer's.

Larry Holtz
KGON/Apogee Communications
Portland, Oregon

(KGON was sold to Apogee Communications in October 1992.)

Richard Tell
Richard Tell Associates, Inc.
Las Vegas, Nevada

Eric Dye &
Ali Mahnad
JAMPRO Antennas
Sacramento, California

Benjamin F. Dawson III
Hatfield and Dawson
Seattle, Washington

Steve Gerber &
Kevin Kohnstamm
City of Portland
Bureau of Planning
Portland, Oregon

Robert Surette
Shively Antennas
Bridgton, Maine

I would particularly like to thank Juliette G. Bloomer, my mother, the english teacher, for teaching me not only the difference between a gerund and a copulative verb, but the reason why I wanted to know the difference.

VIDEO COMPRESSION

Wednesday, April 21, 1993

Moderator:
Tony Uyttendaele, Capital Cities/ABC Inc., New York, New York

DIGITAL MULTI-CHANNEL TELEVISION SIGNAL TRANSMISSION SYSTEM
Takayuki Tanaka, Takeo Tsutsui and Ryu Watanabe
NHK Engineering Administration Department
Tokyo, Japan

METHODS FOR MOTION ESTIMATION AND THEIR APPLICATION
Tim Borer
Snell & Wilcox
Petersfield, Hampshire, U.K.

***TESTING VIDEO QUALITY OF DIGITAL TELEVISION CODECS**
G. W. Beakley, C. P. Cressy & E. S. Kohn
StellaCom
Arlington, Virginia

*Paper not available at the time of publication.

DIGITAL MULTI-CHANNEL TELEVISION SIGNAL TRANSMISSION SYSTEM

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Tokyo, Japan

1. ABSTRACT

NHK has developed a digital multi-channel TV signal transmission system based on its newly developed bandwidth compression technology advantageous to the transmission of the NTSC signals. Efficient bandwidth compression is obtained through Y/C separation, followed by color subcarrier conversion into the amplitude and phase data, then through subjecting each of them separately from the Y signal to the DCT process. This paper shows the superiority of this system in principles, and also reports on a digital 3-channel multiplex FPU unit constructed for the evaluation of the system.

2. BACKGROUND

Now, it is widely accepted that the transmission lines of television signals should be constructed digitally. However, for the effective utilization of the transmission lines, it is necessary, not to transmit digital television signals as they are, but to transmit them with their data rate reduced by any measure. For this purpose, various technologies for bandwidth compression and multi-channel transmission of television signal data have been studied. In this paper, this problem is discussed with the NTSC signals of the current standard television system in Japan and America.

In realizing the digital transmission and multi-channel transmission of the current television signals, it is desirable to construct systems that allow the NTSC signals to be transmitted as they are so that the existing home receivers and station equipment can be utilized. However, if the NTSC signals are compressed while they are composite signals, adequate bandwidth compression cannot be obtained because the color subcarrier contains much high-frequency components and therefore interferes.

The reason will be clarified if, for example, the discrete cosine transform (DCT) process representative of the current data compression systems is taken. In the DCT process, the data rate is reduced by transforming the signals to be transmitted in a manner that the data are concentrated in a region near the DC components. In the case of the composite signals, however, the data spread in the DCT process, not only in the region near the DC components, but also up to a region near 3.58 MHz which is the frequency of the color subcarrier. Thus, the data rate cannot be reduced adequately.

In this paper, it is proposed to first subject the NTSC signals to Y/C separation, then convert the color subcarrier into the polar coordinate components, and subject each of the amplitude and phase data separately from the Y signal to the DCT process. Hereinafter, this technology is called "A-Φ TRANSMISSION SYSTEM". Its principles are described in detail in 3.

3. A − Φ TRANSMISSION SYSTEM

First, how to calculate the amplitude and the phase of the color subcarrier will be explained. Fig. 1 shows the waveform of the color subcarrier obtained through Y/C separation of an NTSC signal. The sampling frequency is taken as 14.3 MHz, which is four times as large as the frequency of the color subcarrier, f sc = 3.58 MHz. Now, from four sampling values, $e_q \sim e_{q+3}$, an amplitude A_q and a phase Φ_q of the color subcarrier in this section will be calculated to obtain the following equations.

$$A_q=\{(e_q-e_{q+2})^2+(e_{q+1}-e_{q+3})^2\}^{1/2}/2 \quad (1)$$

$$\Phi_q=(q \bmod 4+1)\pi/2 - \sin^{-1}\{(e_{q+1}-e_{q+3})/2A_q\} \quad (2)$$

Subsequently, it is only necessary to proceed until A and Φ for all qs are obtained. The sampling frequency of A and Φ obtained in this way becomes 3.58 MHz. However, since the color component bandwidth of the NTSC signal is 1.5 MHz for one color difference signal I and 0.5 MHz for other color difference signal Q, it follows that each of original es can be completely restored according to the sampling theorem.

Then, the meanings of these A and Φ in principles will be explained. First, the relationships of A and Φ with I and Q are as follows.

$$A^2 = I^2 + Q^2 \quad (3)$$
$$\sin\Phi = I/A \quad (4)$$

$$I = A\sin\Phi \quad (5)$$
$$Q = A\cos\Phi \quad (6)$$

$$\Delta Ia \fallingdotseq \Delta A\sin\Phi \quad (7)$$
$$\Delta Qa \fallingdotseq \Delta A\cos\Phi \quad (8)$$

$$\Delta I_{phi} \fallingdotseq A\Delta\Phi\cos\Phi \quad (9)$$
$$\Delta Q_{phi} \fallingdotseq -A\Delta\Phi\sin\Phi \quad (10)$$

At this point, the error power in the A−Φ transmission is caluculated as follows.

$$\Delta Ia^2+\Delta Qa^2+\Delta I_{phi}^2+\Delta Q_{phi}^2 \fallingdotseq (\Delta I^2+\Delta Q^2)\{1+\Delta I^2/(I^2+Q^2)\} \fallingdotseq \Delta I^2+\Delta Q^2 \quad (11)$$

This indicates that, in respect of the error power, there is no significant difference between the A−Φ transmission and the I−Q transmission.

The quantizing error of the hue, ΔΦ, is calculated as follows.

$$\Delta\Phi = -I\Delta Q/A^2+Q\Delta I/A^2 \quad (12)$$

$$\Delta\Phi_{max} = S/A \times 40.5^\circ \quad (13)$$

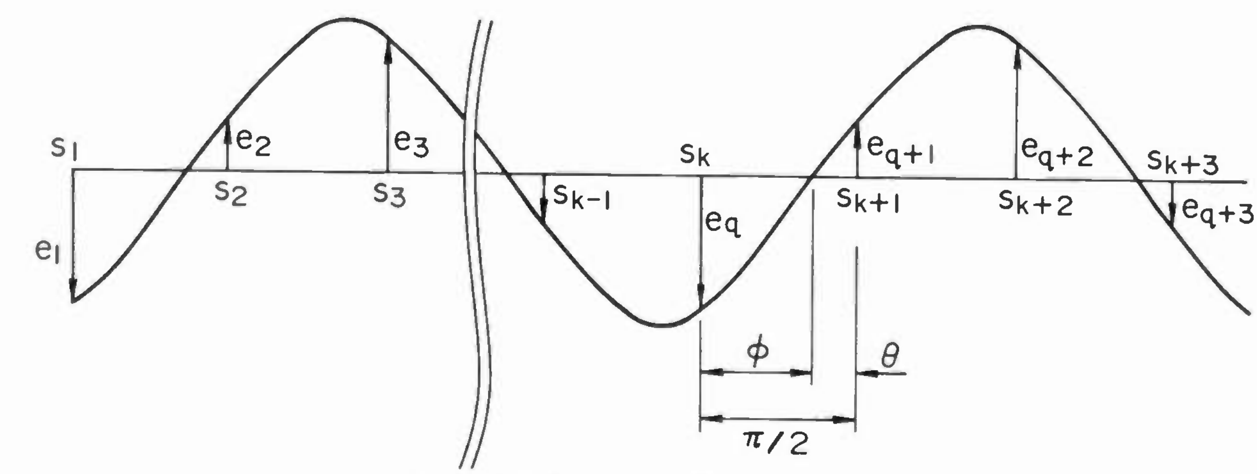

Fig. 1 Waveform of Color Subcarrier

where S is the quantizing step. If quantizing by 8 bits is assumed, then S is about 7.14 mV. This indicates that the maximum quantizing error of the hue in the I-Q transmission does not fall below 1° unless the amplitude of the color subcarrier approaches 290 mV. The distortion of the hue is liable to be very conspicuous to the eye, and therefore, it degrades the picture quality in the subjective evaluation. This is a drawback of the I-Q transmission. In contrast, in the A-Φ transmission, the quantizing error of the hue can be held within a certain range irrespective of the amplitude of the color subcarrier. This is the advantage of the A-Φ transmission.

To conclude this Chapter, the advantages of the A-Φ transmission over the I-Q transmission can be summarized as follows.

a) There is no picture quality degradation caused by repeated demodulation and modulation of I and Q.

b) There is no need for considering sampling axes as in the I-Q transmission.

c) The data rate in transmitting amplitude data is smaller than that required to transmit one of color difference signals. For example, if quantizing by 8 bits is assumed, then I and Q vary between -128 and +127, while A remains within the range of 0 to 181.

d) If the amplitude of the color subcarrier is within the range of normal television signals, then the maximum quantizing error of the hue is smaller than that in the I-Q transmission. For example, if Φ is transmitted at 8 bits, then $\Delta\Phi max$ is 1.4° and therefore, the A-Φ transmission is advantageous over the I-Q transmission except when A is above about 200 mV.

4. THREE-CHANNEL MULTIPLEX DIGITAL FPU EQUIPMENT

This Chapter will introduce an FPU equipment experimentally manufactured based on the A-Φ transmission system. This equipment is intended to transmit three channels of NTSC signals added with stereophonic audio signals in a 18 MHz bandwidth. Each of the transmitting section and the receiving section consists of a single rack measuring 570W × 1700H × 600D mm. Table 1 shows the rated values.

Number of channels	Video: 3 Audio: 6
Base band signal bit rate	Video :(average) 5 Mbps/channel Audio :(average) 128 kbps/channel Control:2 Mbps
Error correction	Reed-Solomon (255,239) Convolution (3/4)
Modulation bit rate	25.47 Mbps
Type of modulation	MSK

Table 1

4.1 Encoding Unit

Fig. 2 shows the functional block diagram of the encoding unit. For all the three channels, the same configuration composed of a single shelf plate measuring 480W × 400H × 600D mm is adopted. By reference to Fig. 2, the functions of the encoding unit will be explained.

First, the NTSC signal is input to this unit. It is quantized by 10 bits at the clock frequency of 14.3 MHz into a digital signal. Then, Y/C separation of movement adaptive type takes place. The C signal of 10 bits is converted into amplitude data A and phase data Φ, each of 8 bits by the system described in Chapter 3.

On the other hand, the Y signal is input to the DCT circuit of 8 × 8 fixed block size and then to the quantizing circuit. Here, the quantizing table is not a fixed one, but is a 16-step variable table of steps 0 to 15 controlled by a rate buffer control circuit described in detail in Chapter 5. As standard, a level-7 table as shown in Table 2 is used. Quantizing becomes finer as the level approaches 0, or becomes coarser as the level approaches 15. The quantized data is input to the Huffman encoding circuit where a variable length code is formed, which is then output through a rate buffer at a certain constant bit rate.

One of the quantizing circuit outputs is decoded through a local decoder composed of an inverse quantizing circuit and an inverse DCT circuit and is input to a frame memory. The frame memory is used to detect movement vectors and also adaptively to do prediction switching between intra-field, inter-field and inter-frame.

Horizontal

Vertical	1	2	3	4	5	6	7	8
1	1.0	1.0	1.0	1.7	2.9	5.3	7.8	10.0
2	1.0	1.0	1.0	2.0	3.2	7.1	12.0	15.0
3	1.0	1.0	1.2	2.4	3.5	8.5	16.0	21.0
4	1.0	1.0	1.3	2.6	3.9	10.0	17.0	25.0
5	1.0	1.1	1.4	2.9	4.7	11.0	21.0	27.0
6	1.1	1.4	1.7	4.3	6.9	13.0	22.0	29.0
7	1.3	1.8	2.1	5.6	9.1	17.0	24.0	31.0
8	1.6	1.9	2.3	6.8	11.0	19.0	28.0	33.0

Table 2

Circuits for the A and Φ data function similarly to the case of the Y signal, but the movement vector detection is not done and Y's movement vectors are utilized.

Thus, all Y, A and Φ are input to the rate buffer and ultimately 5 Mbps outputs are provided.

4. 2 Decoding Unit

Fig. 3 shows the functional block diagram of the decoding unit. For all three channels, the same configuration composed of a single shelf plate measuring the same dimensions as the encoding unit is adopted. From the number of circuits, it is naturally expected that this unit can be constructed in a size considerably smaller than the encoding unit. However, because of the experimental manufacture, it is given the same dimensions as the encoding unit. The number of mounted PCBs is different, of course. Basically, each circuit of the decoding unit is inverse in its function to the corresponding circuit in the encoding unit, and will not be repeated here.

4. 3 Modulation and Demodulation Units

Fig. 4 and 5 show the block diagrams of the modulation and demodulation units, respectively. Table 3 shows their specifications. Both units are constructed with a common shelf measuring 480W × 150H × 370D mm.

5. RATE BUFFER CONTROL SYSTEM

In Chapter 4, the configuration of the digital 3-channel multiplex FPU equipment was described with the exception of its 3-channel multiplexing unit, which will be described in this Chapter. This unit employs a novel control system as shown below.

Generally, in the conventional multiplexing units, the bit rate for each channel was fixed, for example, at 5 Mbps. However, since each channel has different picture entities usually, its redundancy is naturally different. For example, between sports pictures of active movements and landscape pictures of natural scenes, redundancy is obviously higher for the latter. If the same bit rate is allocated in spite of such differences in redundancy, then the picture quality will be badly degraded with the sports picture whereas it will be less degraded than necessary with the landscape picture. For this reason, the multiplexing unit of this equipment has been constructed to permit a bit rate allocation in accordance with each channel redundancy.

Concretely, from the rate buffer control section of each channel, write and read addresses are input to the multiplexing unit, where the empty status of the buffer of each channel is monitored by comparison of both addresses, and the reading rate of each channel is adjusted so that the status of the buffer is equalized for all channels. Moreover, the quantizing table of each channel is corrected of its level. In this way, in the multiplexing unit of this equipment, the transmission bit rate of each channel is adaptively controlled to approximately uniform picture quality degradation by compression of each channel.

Transmitting Unit	
Transmitting bit rate	21.47Mbps(6 f sc)
Transmitting bit rate stability	±250 bps
Type of modulation	GMSK
Input signal: Input level Input polarity Input impedance	 ECL(High:- 0.9V, Low :-1.75V) Active H 100 ohms
Input clock: Input level Input polarity Input impedance	 ECL(High:- 0.9V, Low :-1.75V) Positive-going 100 ohms
Output signal: Output level Output polarity Output impedance	 1 Vp-p(10±2dBm) Active H 50 ohms
Carrier frequency Frequency stability Transmission spurious	140 MHz 5×10^{-5} -60 dB

Table 3-a

Receiving Unit	
Input signal: Input level Input polarity Input impedance	 200 mV Active H 50 ohms
Output signal: Output level Output polarity Output impedance	 ECL(High:- 0.9V, Low :-1.75V) Active H 100 ohms
Output clock: Output level Output polarity Output impedance	 ECL(High:- 0.9V, Low :-1.75V) Positive-going 100 ohms
Method of detection	Costus tuning
Bit error rate Eye aperture characteristic	10^{-3} (CN 11.5dB) 80 % max.
Carrier pull-in width Carrier pull-in time Clock pull-in width Clock pull-in time	± 4.5 MHz 1 sec ± 700 Hz 1 sec

Table 3-b

6. CONCLUSIONS

This paper proposed a system suitable for the NTSC signals to be transmitted. The luminance signal, and the amplitude and phase data of the color subcarrier, are separately DCT processed. Based on this system, a multi-channel digital FPU equipment capable of transmitting 3 NTSC channels and 6 audio channels in a 18 MHz bandwidth was experimentally manufactured. The multiplexing unit of this FPU equipment employs a new system to adaptively control the bit rate of each channel according to its redundancy. This FPU equipment, which was experimentally manufactured, has as such problems to be further studied including field test. Present studies of the quantizing table and the Huffman code are also not exhaustive as to whether they are best suited for this system or not.

The existing problem of the greatest importance is how to transmit the phase data of the color subcarrier when its amplitude is small. In any region hardly colored where the amplitude is near zero, even if the phase data is faithfully transmitted, it will merely result in an increased bit rate, which is insignificant for improving the feeling of visibility. At present, it is practised that when the color amplitude falls below a certain limit, a data of a fixed phase should be transmitted, and better methods are being sought. A study is also being made to determine whether the adaptive prediction and the DCT process is really suitable for the phase data transmission at all, or the vector quantizing or any other means may be better.

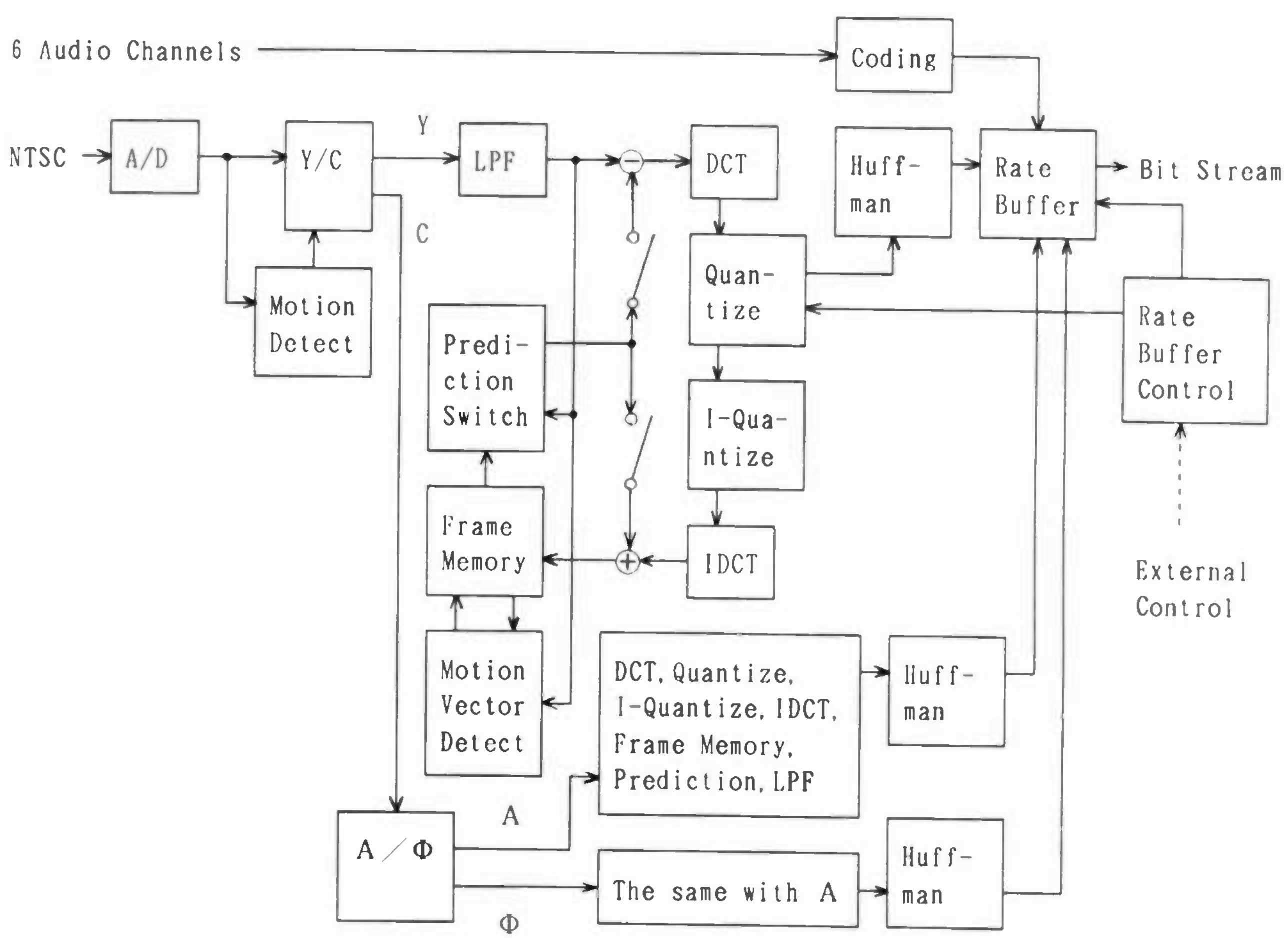

Fig. 2 Encoding Unit

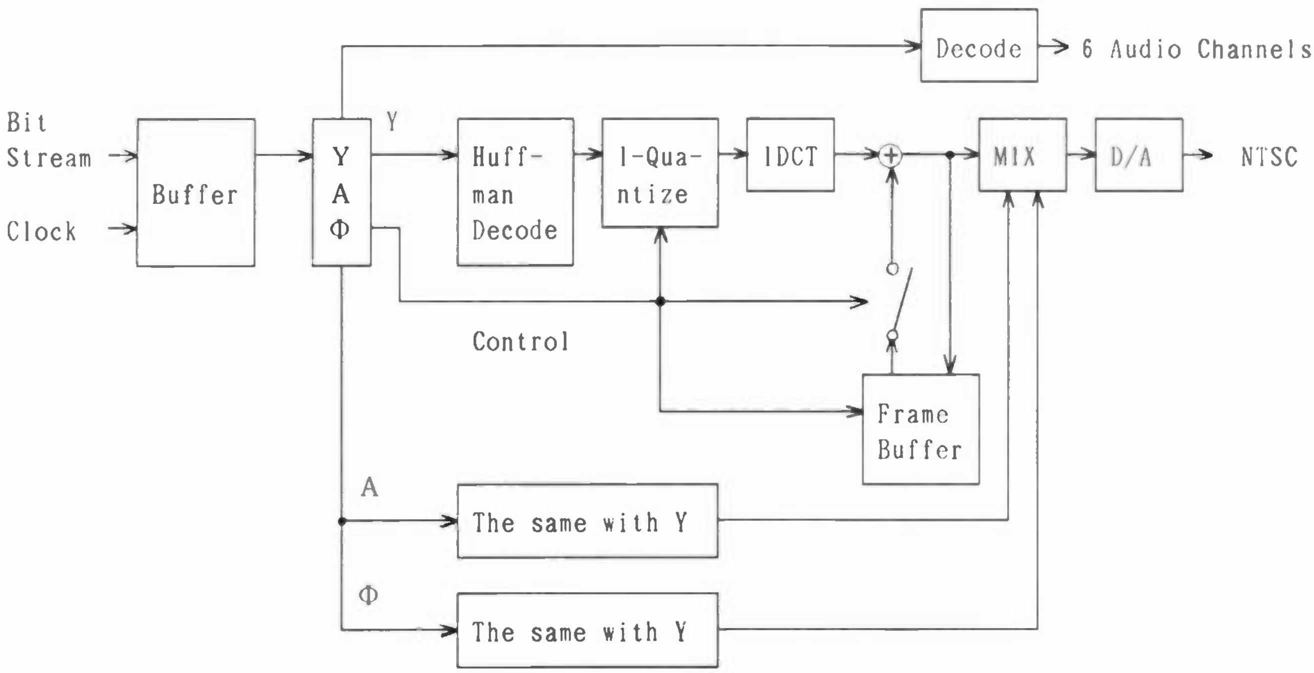

Fig. 3 Decoding Unit

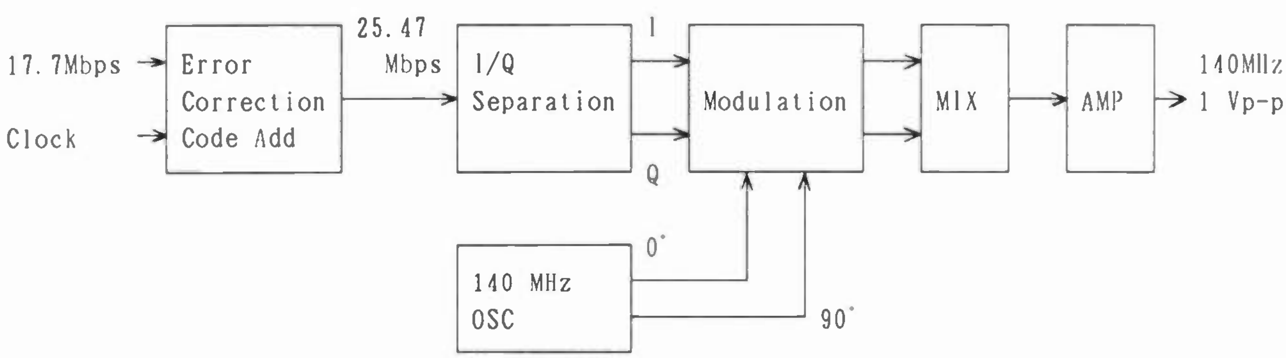

Fig. 4 Modulation Unit

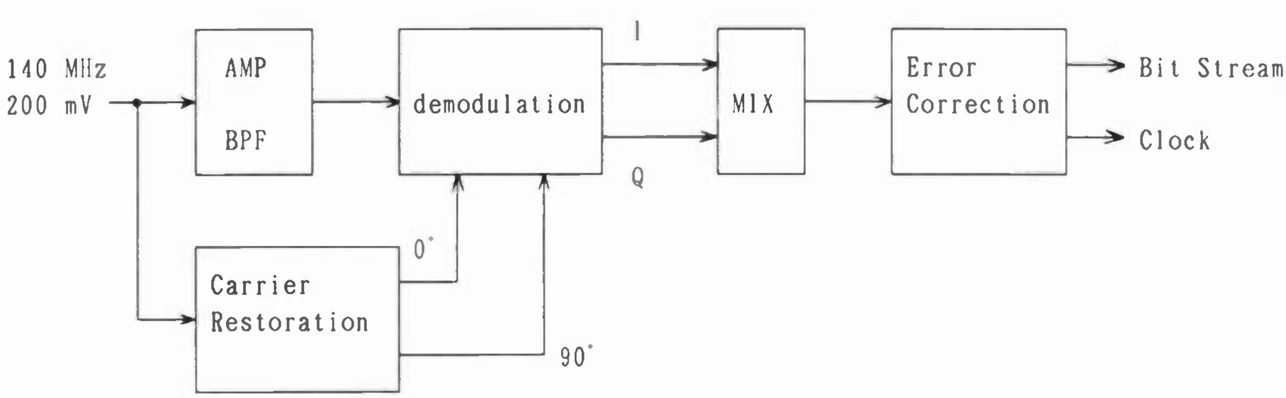

Fig. 5 Demodulation Unit

METHODS FOR MOTION ESTIMATION AND THEIR APPLICATION

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ABSTRACT

This paper describes motion estimation and the applications for which it can be used. The emphasis is on a comparison of the various options available for motion estimation and their suitability for different applications. In particular the differing requirements for video compression and studio applications (with standards conversion as an example) are discussed. Conclusions are drawn about the types of motion estimator which are suitable for these applications.

INTRODUCTION

Motion compensated processing and motion estimation has received much attention in recent years. The technique can be used for many different applications and, in particular, promises improved video compression techniques and artifact free processing of television pictures. This paper discusses the ways in which motion estimation can be performed and the suitability of the techniques for various applications. It attempts to show that motion estimation and its application should be considered together rather than in isolation.

The paper starts by considering the need for motion compensated processing. It discusses the advantages which are possible through this technique and which cannot be achieved in any other way. The applications of motion estimation are discussed. The type of application which can benefit from motion compensation is considered and its benefits in these applications. The relative difficulty of applications is also considered. The techniques of motion estimation are discussed and their suitability for various applications considered.

THE NEED FOR MOTION COMPENSATION

Until the late 1980's algorithms for processing moving pictures in real time were highly constrained by the available semiconductor technology. Therefore conventional processing of video signals (eg. for standards conversion) is often based on linear filtering techniques which are relatively simple to implement. The shortcomings of these conventional processing techniques are well illustrated by the example of standards conversion between European and American television standards. Conventional standards converters essentially try to estimate output pictures using a suitable mix of, perhaps, 4 successive input pictures. This works well for slow movement but results in judder or multiple imaging, and blurring for fast moving objects.

The problems of conventional video processing arise because television pictures are sampled insufficiently often at only 25 or 30 frames per second. Even moderate movement in television pictures can result in large brightness differences between corresponding pixels in consecutive pictures. In the frequency domain these problems are manifest as severe aliasing of the signal. When aliasing is present in a sampled signal (such as television) the assumptions underlying the theory of linear filtering are violated. So it is unsurprising that, when aliasing is generated by motion, conventional standards converters, based on linear filtering, perform less than perfectly.

Curiously, although television pictures are significantly undersampled temporally, nevertheless temporal aliasing is seldom perceived by the viewer. Temporal aliasing only becomes apparent when the image has been processed in some way. The absence of perceived temporal aliasing is because the eye interacts with the moving scene as an 'active' sensor. In particular the eye moves to follow ('track') the movement of objects, thereby keeping their image stationary on the retina. The response of the human visual system to moving television images can be explained in more detail using frequency domain analysis. The interested reader is referred to Tonge '86, Thomas '91 or Borer '92.

Essentially the eye does not see aliasing in unprocessed television pictures because it performs motion compensation of moving objects. The eye tracks the movement of objects and so keeps their images stationary on the retina. By tracking movement the human visual system is able to treat a moving object as if it were stationary. This allows the eye to see much more detail when things move. Conventional standards conversion treats the television image as a whole and each part is processed in exactly the same way. The human visual system, by contrast, treats each object differently depending on its motion.

Motion compensated processing attempts to mimic the operation of the human visual system by treating each object according to its motion. By this means motion compensated processing can achieve the same picture quality on moving objects as conventional processing can achieve on stationary ones. In standards conversion, for example, motion compensated processing can avoid the artifacts cause by aliasing in conventional standards conversion. Therefore standards converted pictures produced using motion compensation are free from judder, multiple imaging and blurring.

There is an apparent paradox in the preceding discussion. The presence of aliasing in the television signal implies that Nyquist's sampling criterion has been violated. If this is the case signal processing theory suggests that there is insufficient information in television signals to perform artifact free processing. The previous paragraphs, however, suggest that motion compensation allows artifact free processing of aliased television signals. How is this possible?

The mechanism by which the limitations of Nyquist sampling theorem can be avoided is by using information about the movement of objects. The motion information must, however, be measured from the original, aliased, scene. Motion estimation is only possible by making certain, a priori, assumptions about television signals. For example, it is often assumed that the scene consists of rigid linearly moving objects. The assumptions necessary for motion estimation are discussed in more detail below. The solution to the apparent paradox is that we are using additional information about television signals (in the form of various assumptions) as well as the signal itself to achieve more than a simple interpretation of Nyquist's sampling theorem would allow. If the television signal does not conform to the assumptions made, for example if the signal were pure noise, then motion estimation will not work. Fortunately television signals almost always obey the highly general and non-specific assumptions which are necessary for motion estimation.

The continued development of semiconductor technology now permits the use more complex processing algorithms than were possible hitherto. One such technique is motion compensation which allows us to avoid the defects which aliasing causes in conventional linear filtering processing techniques. Hence, for example, virtually transparent standards conversion between European and American television is now possible using motion compensation.

APPLICATIONS OF MOTION COMPENSATION

Motion compensation is a very general technique which can be used for virtually any type of processing involving moving pictures. Thus motion compensated processing will gradually come to have applications in many areas of television and computer graphics. Motion compensation requires reliable motion estimation, which is a difficult process which has taken the human visual system millions of years of evolution to develop. Electronic motion compensation is a relatively new technique dating from the late 1970's (Netravali & Robbins '79). Real time motion estimation hardware has only been available for a few years (Borer 90, Thomas '90, Weiss '90). Some years of development will be required to fully refine and reduce the cost of motion compensated equipment.

Although in principle motion compensation can be used for most types of television processing some applications are more suitable than others. Because of the complexity of motion compensated processing there must be a clear advantage to using it. For example, motion compensation could be used converting between conventional TV and HDTV (at the same field rate), that is for line rate up and down conversion. However, high quality can be achieved in these applications without using it (Weston '88, Devereux '92). Motion compensation is most useful in applications which involve a change of field rate, for example intercontinental standards conversion, field rate upconversion (Borer '90, Fernando '88) and slow motion display (Thomas '90, Lyon '92).

Not all applications of motion compensation are equally difficult. Considering again the 3 examples from the previous paragraph; high quality motion compensated field rate upconversion appears to be the least difficult to achieve. Presumably this is because any processing artifacts are present only on one field for a very short period of time. This sort of artifact is difficult for the human visual system to detect. Slow motion display is, perhaps the most difficult application of motion compensation. Any processing artifacts may be present on the display for many fields making them very easy to detect. Hence this type of system is an excellent technology driver for motion compensated processing, allowing the technique to be developed further. In spite of the difficulty of slow motion display the high quality demonstrated by the BBC/Snell & Wilcox slow motion system shows that the benefits of motion compensation can be achieved in practice.

The applications of motion compensation should not be considered in isolation from the method of motion estimation. The application largely defines the requirements of the motion estimator and determines the interface between estimator and application. For example most motion estimators generate motion vectors on the same sampling lattice as the input video, ie with a 525/60 input the motion vectors are generated at 525/60. For some standards converters the motion vectors may be required on a different, output, standard (eg 625/50) rather than on the input standard. Hence in this case the motion vectors themselves may have to be standards converted to the output standard before they can be used (eg Nowak '90, Borer '91). By contrast the motion compensated 'Alchemist' standards converter made by Snell & Wilcox has been specifically designed, in parallel with the motion estimator, to use motion vectors generated on the input sampling lattice. Therefore standards conversion of the motion vectors themselves is not required in this converter.

Motion compensation is a general technique that is applicable to many types of television processing. Motion compensated processing mimics the operation of the human visual system and treats moving objects as though they were stationary. The use of motion compensation avoids loss of resolution (blurring) and aliasing artifacts (eg judder, multiple imaging, flicker) commonly caused by non-motion compensated processing. Motion estimation is the enabling technology which is the key to high quality motion compensation.

MOTION ESTIMATION

Motion estimation is the process of constructing a description of a moving picture, in terms of objects in the picture and their motion. Motion information cannot simply be computed from the image data since, even under ideal conditions, there may be many motion fields which are compatible with the data. The problem of motion estimation, therefore, is to find an appropriate motion field, suitable to the application at hand. It might be assumed that the 'best' motion field would be that closest to the true motion field. This is not necessarily true. The human visual system 'expects' to see motion that is compatible with real world motion. Thus a motion field which is somewhat in error, but realistic, may be preferable to one which is closer to the true motion, but inconsistent with the real world. Furthermore care must be taken in choosing a definition of 'closest' to the true motion. Some motion estimation algorithms can be shown to converge to the true motion *on average*. However if the motion vectors are noisy, the motion field will be disjointed and unrealistic. This type of motion estimator would be unsuitable for applications such as standards conversion.

The velocity field may be defined as the instantaneous velocity of a point, in the image plane, as a function of position and time (ie. $v(x,y,t)$). These velocities are related, through a perspective transformation, to the velocities of objects in the original (3 dimensional) scene. For temporally sampled images (as in television) a function related to the velocity field is the displacement field. This establishes a correspondence between points in the current frame and points in either the previous or subsequent frame. These two displacement fields may be referred to as the backwards and forward displacement fields respectively (eg Cafforio '90). The displacement field is only defined for a specific pixel if the corresponding pixel exists in the previous or subsequent frame. The forward and backward displacement field are thus different, since backward displacement is not defined for regions which have been newly revealed, and forward displacement is not defined for regions about to be obscured. It is usually the displacement, rather than the velocity, field which is used for motion compensated image processing. The difference between the forward and backward displacement fields is significant since it is directly related to regions of revealed or obscured background. The more general term motion field can refer to either a velocity or a displacement field.

In order to perform motion estimation all algorithms make certain, a priori, assumptions about the image. An image is formed by the projection of a 3 dimensional scene on to 2 dimensions. For motion estimation it is assumed that the scene comprises of a number of moving objects, and that each point in the image corresponds to a unique point on the surface of an object. This model is violated in a number of cases, for example with transparent, diffuse (eg smoke) and reflecting scene components. A relationship must also be assumed between the observed image and the unknown motion field. The assumption usually made is of the constancy of image features along the motion trajectory. This assumption is known as the structural model (Dubois '90). A number of different image features could be used, most commonly the image brightness (luminance). Another commonly used feature is the filtered luminance, for example the gradient of the image might be used. Using the image gradient, rather than intensity, improves the validity of the structural model in the cases where there are changes of illumination or shadow, because it removes the zero frequency component of the image. More generally a 3 component colour vector can be taken as constant along the motion trajectory (Dubois ICASSP '90).

The structural model is not, in itself, sufficient to uniquely define the motion field. Even in ideal circumstances the structural model may be satisfied by many different motion fields. In practice the motion field is even less well defined due to noise and inadequacies in the structural model. Therefore an additional assumption must be made about the motion field. This is usually that the motion field is generally smooth, with discontinuities at (spatio-temporal) object boundaries. This assumption is occasionally explicit (eg Horn '81). More often it is implicit in the algorithm (eg in the use of post processing of the motion field (Weiss '90) or in the use of trial vectors in phase correlation (Thomas '87)).

Different applications impose different requirements on the motion estimation process. These will be considered from the point of view of European/American standards conversion. First the relevant parameters will be considered, and then discussed in more detail in subsequent paragraphs. One such parameter is the resolution required of the motion field. That is, how many pixels each measured motion vector corresponds to. The necessary range and resolution of the motion vectors must also be considered. These are the largest velocity likely to be encountered and the precision to which motion vectors are measured. The characteristics of the input images, from which the motion must be estimated, are also important. Finally, since all motion estimation algorithms sometimes fail, the conditions under which an algorithm fails, and its behaviour in these circumstances should be considered.

Ideally for motion compensated standards conversion a distinct motion vector is required for each image pixel. That is the resolution of the motion field should be the same as the original image. This is in contrast to video compression (bitrate reduction) applications, where such a high resolution motion field is not required and, indeed, would be impractical because of the large bit rate required for its transmission. A high resolution vector field for standards conversion allows the precise segmentation of the input image into regions with different velocities (ie different objects). Precise segmentation is essential to produce high quality interpolations. In practice it may be impossible to obtain the motion field to the same resolution as the input image. This may be due to imperfections in the input images (eg from vertical/temporal aliasing caused by interlace). Therefore, in practice, it may be necessary to use a lower resolution motion field, which must then be interpolated to the resolution required for motion compensated processing.

The motion field must be estimated in a way that will allow the range of velocities likely to be encountered, to be measured. Again standards conversion is significantly different from video compression. These algorithms typically only require relatively small velocities (eg ±8 pixels/field period). For larger velocities bit rate reduction algorithms generally switch to an alternative mode of operation (eg spatial subsampling with intra field interpolation). For standards conversion there is no good 'fallback' option for interpolation and large velocities are precisely those for which motion compensation is most needed. Therefore standards conversion applications require the measurement of relatively large velocities (*at least* ±32 pixels/field period). This difference in the range of velocities required can have a significant effect on the motion estimation algorithm used. For example, the 'block matching' algorithm, often used for bit rate reduction, does not scale well to the larger velocities required for standards conversion.

It is difficult to assess the precision to which motion vectors should be estimated. That is how many fractional accuracy bits are need to describe the vectors. Girod and Thoma (December '85) have shown that a signal containing translatory motion can be perfectly interpolated using integer pixel displacement vectors. This sets a lower limit on the required resolution (ie the velocity need only be resolved to the nearest pixel/frame period). However this result is only applicable for pure translatory motion and assumes an infinite number of input images can be filtered. In practice neither of these conditions are met and accuracy to better than integer pixel displacements is required to maintain the full resolution of the input in the interpolated images. The size of the (motion compensated) interpolation aperture used and the nature of source images (eg resolution, interlace etc) both influence the accuracy to which the vectors should be measured. For typical television images the resolution of the image decreases with object velocity because of the camera integration time (this effect should not be over estimated since integration time can be very short with modern CCD cameras). Therefore the measurement accuracy of the motion vectors can decrease with speed. A constant fractional error would probably be appropriate. For 'typical' (interlaced) images, and an interpolation aperture including only a few fields, a vector accuracy of, perhaps, ¼ or ⅛ pixel/field period might be appropriate at low velocities.

Finally, from a practical point of view, it should be remembered that motion estimation can be computationally expensive. Therefore, in order to produce real time hardware, it may be necessary to use a sub-optimal algorithm to achieve a practical hardware complexity. The practical limit to hardware complexity depends on the application. Fortunately, high quality standards conversion in the television studio can probably afford to be more complex than in other applications (eg bit rate reduction for teleconferencing).

Motion estimation algorithms may be broadly classified into 3 groups, spatio-temporal constraint techniques (gradient methods), region matching methods (block matching) and frequency domain techniques (eg phase correlation). These are discussed, in turn, below. In order to minimise the computational complexity, the first two methods are often implemented recursively, as described below. Motion estimation techniques can also be applied in a hierarchical manner to deal with different object sizes and large velocities.

The use of iterative (or recursive) techniques is common in both region matching and constraint techniques. The basis of the technique is to use a motion estimation technique to improve the accuracy of an initial velocity estimate. The new estimate can then be used as the basis for another iteration to improve accuracy further. For example, given an image g(x,y,t) and an initial estimate of the velocity (u_0,v_0) an 'improved' estimate can be made by applying the motion estimation algorithm to $g((x+u_0t),(y+v_0t))$. That is, a motion estimate can be improved by compensating for the initial velocity estimate before performing further motion estimation. This gives a new velocity estimate (p_1,q_1). The 'improved' velocity estimate is then given by (u_0+p_1,v_0+q_1). The process can be summarised as follows. Let **ME** be a motion estimation operator which acts on a region of a moving image to give an estimate of the motion ie:

$$ME(g(x,y,t)) \Rightarrow (u,v) \tag{1}$$

Then the iterative process can be described as;

$$ME(\ g((x+u_nt),(y+v_nt),t)\ ) \Rightarrow (p_{n+1},q_{n+1})$$

$$(u_{n+1},v_{n+1}) = ((u_n+p_{n+1}),(v_n+q_{n+1})) \tag{2}$$

A variety of 'recursive' techniques arise from different choices of the initial estimate and the number of iterations to be performed. If initial velocity estimate is zero and iteration is performed as described above, the process is known as motion compensated iteration (Bierling '86). This is often used with spatio-temporal constraint techniques. Alternatively either spatial or temporal recursion may be performed (Netravali '79, Paquin '83). In these techniques the initial estimate is taken from an adjacent pixel, either spatially or temporally. Both spatial and temporal recursion risk error propagation at (spatial or temporal) object boundaries. For spatial recursion the accuracy of the motion estimate also depends on the relative directions of object motion and the recursion (Robert 1985). This problem can be avoided by performing spatial recursion in 2 (or more) directions simultaneously and selecting the best result (Cafforio '90, Hann '90). Unfortunately it is difficult to guarantee the convergence of motion compensated, spatial and temporal recursive schemes.

In hierarchical processing, motion is estimated at a number of different image scales. Typically a pyramid of different image resolutions is used. The lowest level of the pyramid is the original image and successively higher levels in the pyramid are obtained by low pass filtering the lower level image (perhaps with subsampling to reduce computation). Large regions of the image are examined, in low resolution, at the top of the pyramid, while small regions are examined, in high resolution, at the bottom.

One way of implementing hierarchical motion estimation is as a form of recursion. Hierarchical motion estimation, generally, uses information from the image at a variety of different scales. This is achieved by applying the motion estimation algorithm to different sized regions of the image to extract the maximum information. Usually the larger regions are filtered and subsampled to minimise computation. One form of hierarchical motion estimation uses the recursive technique described above (equations 1 & 2). The initial velocity estimate comes from a larger, lower resolution, region of the image (Bierling '86, Martinez '87). The largest, lowest resolution, image region used would typically take zero as the initial velocity estimate. This hierarchical technique allows the measurement of large velocities with high accuracy and produces a high resolution (dense) velocity field. With this form of motion estimation the maximum measurable velocity depends on the resolution of the lowest resolution image used. However, with low resolution images it may not be possible to detect the movement of small objects. There is, thus, a relationship between the highest velocity which can be measured and the smallest object which can be detected.

Consider the image feature (eg picture brightness) represented by the function g(x,y,t). For motion estimation the image is assumed to represent a moving object. Hence the image brightness, g, may be expressed as,

$$g(x,y,t) = h(\alpha(x,y,t),\beta(x,y,t),t) \tag{3}$$

where h is the brightness of the *object* (at t=0) and α and β are the (horizontal and vertical) coordinates of a point on the *object*. The functions α and β are the spatially varying transformation of the object as a function of time. The dependence of the object brightness, h, on time, allows for changes of illumination etc with time and will, in general, be quite small. Note that this representation of the scene allows for quite general distortions of the object. In the particular case of an object translating with a uniform velocity, (u,v);

$$\begin{aligned} h(\alpha,\beta,t_1) &= h(\alpha,\beta,t_2) \quad \forall\ t_1,t_2 \\ \alpha(x,y,t) &= x-ut \\ \beta(x,y,t) &= y-vt \end{aligned} \tag{4}$$

The first of these equations indicates that the appearance of the object does not change with time, while the latter two equations show that the transformation is a simple translation. For more complex types of motion (eg zooms, rotations, sheers and accelerations) it is possible to formulate more general expressions for α and β, see, for example, Schalkoff '84, Martinez '87 or Wu '90.

Spatio-temporal constraint methods

Spatio-temporal constraint methods are based on the relationship between the gradients of the image brightness. The relationship, between the partial derivatives of the image brightness, can be derived by differentiating equation 3 and using equation 4, ie;

$$\frac{\partial g}{\partial x}=\frac{\partial h}{\partial \alpha}\frac{\partial \alpha}{\partial x}+\frac{\partial h}{\partial \beta}\frac{\partial \beta}{\partial x}=\frac{\partial h}{\partial \alpha} \qquad (5)$$

$$\frac{\partial g}{\partial y}=\frac{\partial h}{\partial \alpha}\frac{\partial \alpha}{\partial y}+\frac{\partial h}{\partial \beta}\frac{\partial \beta}{\partial y}=\frac{\partial h}{\partial \beta} \qquad (6)$$

$$\begin{aligned}\frac{\partial g}{\partial t}&=\frac{\partial h}{\partial \alpha}\frac{\partial \alpha}{\partial t}+\frac{\partial h}{\partial \beta}\frac{\partial \beta}{\partial t}+\frac{\partial h}{\partial t}\\&=-u\frac{\partial h}{\partial \alpha}-v\frac{\partial h}{\partial \beta}+\frac{\partial h}{\partial t}\end{aligned} \qquad (7)$$

from which;

$$\frac{\partial h}{\partial t}=u\frac{\partial g}{\partial x}+v\frac{\partial g}{\partial y}+\frac{\partial g}{\partial t}=0 \qquad (8)$$

This equation is known as the spatio-temporal constraint equation. It relates the partial derivatives of the moving image, when the appearance of the moving object does not change (ie when $\partial h/\partial t=0$). It is *exact* for a scene comprising a uniformly translating object. By using more general equations for α and β it can be generalised to include other motions such as zooms, rotations and sheers (see Martinez '87).

The velocity at an image point can be estimated by minimising the following error function;

$$Error=\iiint_{(x,y,t)\in R}\left(\frac{\partial h}{\partial t}\right)^2 dx\,dy\,dt \qquad (9)$$

where the temporal gradient ($\partial h/\partial t$)of the objects brightness is defined in equation 8. The error is minimised over a region, R, for which the velocity (u,v) is assumed constant. The size of the region must be carefully selected. If it is too small there will be insufficient information to form a reliable motion estimate, if it is too large the assumption of a single velocity may be false. Different region sizes can be used in a hierarchical approach.

A problem with constraint techniques is that they fail at the boundaries (particularly temporal boundaries) between objects. With even with moderate motion, the same pixel in successive frames may correspond to different objects, thus rendering any estimate of the temporal derivative meaningless. Essentially the above techniques are difficult to apply because of the problem of estimating the partial derivatives. These cannot be reliably measured because, at the boundaries of objects, 'the actual image signal differs drastically from the mathematical model the algorithm is based on' (Bierling '86).

The problem of estimating (particularly) the temporal partial derivative of the image stems principally from temporal undersampling of the image. In the context of motion compensated standards conversion it is precisely because of temporal undersampling (and consequent aliasing) that motion estimation is required. The maximum displacements that can be extracted depend on the local frequency content of the image. For example in random dot patterns (white spatial noise) the maximum displacement that can be extracted is one pixel (Fennema '79). Essentially the accuracy of the temporal derivative can only be guaranteed if the velocity is less than 1 pixel per frame (or field). For higher velocities it must be assumed that the source image does not contain the highest frequencies which can be supported by the (spatial) sampling lattice. This assumption may or may not be true, and becomes increasingly unlikely as the motion speed increases.

The problems of constraint techniques lead to the use of recursive (iterative) or hierarchical techniques to improve the range and accuracy of motion estimation. Unfortunately recursive schemes cannot be guaranteed to converge and may not give the 'true' velocity even if they do. With hierarchical techniques, since the process starts with a low resolution version of image, it may not be possible to detect the motion of small objects.

Constraint techniques of motion estimation are typically used for bit rate reduction for teleconferencing. They are quite suitable for this application since expected motion is small. Furthermore, absolute fidelity of the motion field is not required since errors in motion estimation can be corrected by the rest of the bit rate reduction system. The use of a recursive technique yields an effective, computationally efficient algorithm. For standards conversion, by contrast, large velocities may be expected and good motion fidelity is required. To achieve this a hierarchical approach is necessary. This type of motion estimator, for standards conversion applications, would be much more complex than for, say, teleconferencing. Furthermore there is the risk that the motion of small objects may not be detected. Hence this type of algorithm is not ideal for standards conversion.

Region matching methods

Block matching is the best known region matching method of motion estimation. These techniques are based on the assumption that the correct motion transformation will maximise the correspondence between regions in successive images. If this is true then the correct motion will minimise average (modulus) difference between successive displaced images described as;

$$DFD(x,y,t_0) = \iint_{(x,y)\in R} \left|g(x,y,t_0)-g((x-u\Delta t),(y-v\Delta t),t_{-1})\right| dx\,dy \tag{10}$$

DFD is the displaced field or frame difference, g(x,y,t) is the image brightness as a function of position and time and R is a region of the image over which the velocity is assumed constant. $\Delta t=t_0-t_{-1}$. Equation 10 says that, having allowed for the motion in the image, the difference between the two images is entirely due to changes in the appearance (shape, illumination) of the object.The velocity at the centre of the region is defined to be that velocity which gives the minimum error for that region.

Minimisation of the displaced field difference (DFD) is a classic, non-linear, optimisation problem. Other examples of this type optimisation can be found in other branches of image processing eg filter design and image restoration. The (2 dimensional) space of possible velocities must be searched to find a global minimum. This process is complicated by the possible presence of multiple local minima. It is simplified because the search space is only 2 dimensional. The various region matching, motion estimation methods are different methods for solving this minimisation problem, in real time and with practical hardware. The algorithms used can be grouped into 'recursive or gradient' and 'block matching' techniques.

Finding velocity by minimising the displaced field difference can be approached by a number of well known techniques (eg. Stearns '88, Lim '90). For example the steepest descent (Netravali '79) or Newton-Raphson (Lucas '81) algorithms may be used to minimise the DFD. In general these types of algorithm are iterative with each iteration giving an improved estimate. They may be described by the general iterative scheme;

$$v_{i+1} = v_i - \varepsilon R\cdot\nabla\left(|DFD(v_i)|^2\right) \tag{11}$$

where v_{i+1} is an improvement to the estimate v_i and **R** is a recursion matrix (the mean square of the DFD is used as an example). As discussed previously the use of a recursive scheme permits a plethora of algorithms using spatial, temporal, motion compensated or hierarchical iteration.

Care must be exercised when using this type of recursive motion estimation technique. Moorhead et al (1987) have analysed the convergence of Netravali's technique and shown that the *expected value* of the velocity converges to the true value. However the variance of the measured velocity is proportional to the step size ε, which also determines the rate of convergence. Hence there is a tradeoff between the rate of convergence and the noise in the measured velocity at convergence. This type of iterative scheme is often used because of its low computational complexity. For computational simplicity there is usually only one iteration per pixel. Hence a rather noisy velocity field would be expected. In addition to these problems there is the same difficulty in determining the image gradient as for constraint techniques of motion estimation. Thus, used directly, this iterative scheme would only be expected to work for quite small velocities. In order to achieve the velocity range and accuracy required for standards conversion a hierarchical scheme, possibly with multiple iterations per stage, would probably have to be used. These embellishments greatly increase the computational complexity and render the technique less than ideal for standards conversion.

An alternative, straight forward, way to minimise the norm of the DFD is to quantise the search space (u,v) and transform a continuous minimisation problem into a discrete search. This technique is known as block matching. The DFD is calculated for a discrete number of values of velocity. These values are then searched to find the minimum. The velocity must be quantised in integer pixels per frame unless image interpolation is used. The maximum velocity which can be measured is determined by the size of image region used. If the velocity range is doubled, the number of possible quantised velocities is quadrupled (2 dimensions) and the computation per pixel also quadruples. Hence the computational complexity increases as the forth power of maximum velocity. While block matching is a reasonable option for regions of 8x8 pixels, typically used for bit rate reduction, it rapidly becomes impractical for the velocities required for standards conversion. Furthermore a discrete search does not necessarily give an accurate estimate of the velocity because of local minima and aliasing of the DFD signal. A general difficulty with region matching techniques is that their accuracy depends on the frequency content of the picture. If, as is typical for television images, the image spectrum decreases quite rapidly with frequency, it is difficult to accurately measure motion vectors.

<u>Frequency domain methods</u>

The frequency domain properties of moving objects can also be used to determine motion in an image. An example of this is the phase correlation technique of motion estimation. This was first used by Pearson et al ('77) for image registration and later by (Thomas '87) for television motion measurement. Essentially the phase difference between spatial spectra of successive frames is used to determine the velocity of moving objects. Computational efficiency is achieved by using the Fast Fourier Transform (FFT) algorithm to calculate the required spectra.

The phase correlation function of two consecutive images is defined by;

$$\Phi\big(g(t_0),g(t_1)\big) = \mathcal{F}_x^{-1}\left\{ \frac{G(t_0).G(t_1)^*}{\left|G(t_0).G(t_1)^*\right|} \right\} \quad (12)$$

where $\mathcal{F}$ represents Fourier transform and subscript x indicates a spatial transform. An interesting interpretation of the phase correlation function is as the probability distribution of velocities (Girod '89). Calculation of a phase correlation function for every pixel in an image would be too computationally intensive if large image regions are used. Furthermore, the use of large image regions would risk missing the motion of small objects. Hence, a hierarchical approach is used. If there is more than one moving object in the image region used, then multiple peaks can be detected in the phase correlation function. Each peak in the phase correlation surface corresponds to the velocity of a different object. It is assumed that all objects in the image region are moving with one of these velocities. The DFD is calculated for each pixel, for each of the possible motion vectors. The motion assigned to each pixel is that which gives the smallest norm DFD. In this way the image is segmented into regions corresponding to each velocity. Hence a dense motion field can be calculated. Computational complexity for the second stage of the hierarchy is low since the region used in calculating the norm of the DFD is relatively small. More detail on phase correlation can be found in Thomas '87, Dabner '90, Borer '90, Borer '92.

The phase correlation technique of motion estimation has several advantages for standards conversion and similar applications. It is computationally efficient for finding motion vectors if large velocity range and high accuracy is required. This is achieved by taking advantage of the computationally efficiently of FFT algorithms. The accuracy of the technique is very much less dependant on the spectral content of the images. For example phase correlation performs very well on textured images (eg grass or foliage) which cause problems for other motion estimation techniques. Phase correlation is robust with respect to changes in lighting and television fades which are also problematic for constraint based and region matching techniques. Phase correlation also generates an 'error' signal which indicates the confidence which can be place in the motion vector. This allows motion compensated applications to degrade in a graceful way as the motion estimate becomes less reliable (as occurs in all motion estimators). This avoids switching artifacts which can be objectionable in some motion compensated systems. For these reasons phase correlation has been selected by Snell & Wilcox for its television studio equipment.

CONCLUSIONS

This paper has considered the need for motion compensation, the ways in which it can be achieved and the relative merits of the various techniques for different applications. For video compression motion compensation promises improved quality and reduced bit rate. For studio processing of television signals it promises fewer artifacts (such as blurring, judder, multiple imaging and flicker). This is particularly important for the process of European/American standards conversion where processing cannot be avoided and the artifacts of conventional techniques can be both intrusive and objectionable.

For video compression applications constraint based techniques are suitable. They provide acceptable performance in these application with modest complexity. Absolute fidelity of the motion field is not achieved by these techniques but is not required for video compression.

Studio applications, such as standards conversion, have much greater requirements for motion estimation. They require a large range of motion vectors, high spatial resolution, high precision and good fidelity to the true motion. Although constraint based or region matching techniques could be used, they require the embellishments of hierarchical and/or recursive techniques. These significantly increase the cost and complexity of such systems. Therefore for high quality studio processing of television pictures the frequency domain technique of phase correlation is better suited.

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VIDEO COMPRESSION WORKSHOP*

Wednesday, April 21, 1993

Moderator:
Tony Uyttendaele, Capital Cities/ABC Inc., New York, New York

A panel of experts answers questions about compressed video in daily application.

*Papers for this session were not available at the time of publication.

FCC & FAA WORKSHOP*

Wednesday, April 21, 1993

Moderator:
Dane Ericksen, P.E. Hammett & Edison, Inc. Consulting Engineers, San Francisco, California

A panel of FCC and FAA experts address questions concerning issues involving shared jurisdiction.

*Papers for this session were not available at the time of publication.

SATELLITE AND AUXILIARY SERVICES WORKSHOP

Wednesday, April 21, 1993

Moderator:
Fred R. Morton, Jr. KMGZ-FM, Lawton, Oklahoma

A PROPOSAL TO RE-ALLOCATE AURAL RPU FREQUENCIES FOR INCREASED EFFICIENCY
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A PROPOSAL TO REALLOCATE AURAL RPU FREQUENCIES FOR INCREASED EFFICIENCY

Dan Rau
Marti Electronics
Cleburne, Texas

For decades broadcasters have relied on auxiliary broadcast service bands to provide not only remote broadcast capability and studio links but also for basic station communications, IFB and telemetry. With the tremendous pressure of increased usage in not only large markets but medium and small markets as well it's time to reconsider the allocation and use of the RPU frequencies

ORIGINAL AUXILIARY BANDS (37 channels) *(Illustration #1)*

Originally, just three bands were authorized for use by broadcasters, 1600KHz.. with 3 channels, 26MHz. with 25 channels and the 152/153 MHz. with 9 channels shared with industrial users. As the usage increased the 26MHz. band became virtually unusable due to propagation. It was not uncommon for a broadcaster in Texas to pick up a signal from Chicago or Atlanta. Of course with only 3 10KHz. channels the 1600KHz. band didn't last long either. Another mitigating factor to the demise of the 1600 KHz.. band was the growth in the AM broadcast band.

AUXILIARY BAND PLAN IN THE MID 1950's. (60 channels) (Illustration #2)

In the mid 1950's, the 450/455 MHz. band was granted to broadcasters but there were only 16 channels allocated. These were all 100KHz. wide, ostensibly to provide high quality program channels. Ofcourse as broadcasters discovered the joys and profits of remote broadcasts even those channels were quickly filled and the band was once again very congested.

REVISED BAND PLAN 1975 (100 channels) (Illustration #3)

In the early 1970's the band plan went back to the drawing board and was added to and redivided. The 1975 band plan provides for a total of 100 channels. These include the original three 1600KHz. channels, the twenty five 26MHz. channels and the nine 152/153MHz. band. We also have the five 161MHz. channels, one at 166.25, one at 170.15 and fifty six new channels in the 450/455 band. To add more frequencies the FCC authorized more space in the UHF band and dividided the 450/455 band into several different classes of service and bandwidth channels. Channel bandwidths range from 10KHz. to 100KHz.. to provide either telemetry, program, cues or communication.

However, of those 100, 28 are gone due to interference or propagation problems and/or no equipment available. Nine are very limited use due to their being shared with industrial users and two, 166.25 and 170.15 have limited use areas. In reality of the 100 channels available only 61 are truly available for exclusive use by broadcasters and of those 61 eight are assigned for telemetry use only leaving a total of 53 voice quality channels for remote broadcast.

REVISED BAND PLAN OF 1985 WITH SEGMENTS (?? channels) (Illustration #4)

In 1985 the FCC decided to redivide the bands again to provide even more channels. This time making the process of frequency selection totally incomprehensible. The entire set of 152/153, 161 and 450/455 MHz. bands were divided into 5KHz. "segments".

A user is to take as many of these "segments" as they require to make a channel up to the maximum bandwidth authorized in that channel.

(Illustration #5)

For example on this chart you will see how the 161 band has been divided into 5KHz. segments. A user is to pick up to 6 of these segments to make his channel. The problem comes when he selects the channel. Where are the other users in relation to the segments he has picked? If for instance somone picked the first 6 segments, 161.6275, 161.6325, 161.6375, 161.6425, 161.6475 and 161.6525, his center channel would be the current standard channel 161.64.

But what if he picks 6 segments starting at #2, 161.6325, 161.6375, 161.6425, 161.6475, 161.6525 and 161.6575? His center channel becomes 161.6450. That might not be a problem unless there is a user already on 161.64. As you can see, our user would now have a legal channel but be only 5KHz. off an existing channel. This becomes worse if our user starts his 6 segments at 161.6475 and goes to 161.6675. He then will have totally rendered both the current 161.64 and the 161.67 unusable as well as his own new channel if three broadcasters are trying to do remotes at the same time..

We have found the 1985 band plan as outlined in 47CFR Chapter 1, to be not only confusing but unworkable as well. Even the FCC admits that they have no way to properly assign and track the channels in these 5KHz. segments.

It's obvious to broadcast users in medium and large markets that there is severe crowding of the auxiliary bands. Therefore we propose the following band plan.

PROPOSED NEW ALLOCATION, 1993 (124 channels) (Illustration #6)

This plan will give broadcasters a total of 124 25KHz. channels including 3 at 1600KHz., 25 at 26MHz., 9 at 153MHz., 9 at 161/166/170MHz. and 78 at 450/455MHz. There is still the problem with 1600KHz. and 26MHz., but we feel that it is best to not give up any channels even with the known problems.

To assure that everyone will have an opportunity to license and use a relatively trouble free frequency the 450/455 band will be divided into user areas. Of the 78 avaliable UHF channels, 8 will continue as TSL frequencies at 10KHz. bandwidth. The 70 remaining channels will be assigned as follows--18 channels will be used for fixed repeaters and 5 will be reserved for "network operations" on a priority basis. Local entities can license and use the 5 "network frequencies" with the understanding that a network coming to town on a temporary basis will have priority use of the channels. The remaining 55 will be used for remote broadcast functions such as mobile broadcast and portable repeaters.

The new bandplan will result in 85 usable channels and help relieve the severe crowding in the RPU band

All of the channels will be 25KHz. wide. In view of new technology coming to the market place, we feel that high quality RPU transmission in a 25 KHz. channel is a reality. Therefore it won't be necessary for 50 and 100KHz. channels to achieve the music quality remote broadcasts.

Redefining the priority of use.

Adding more frequencies is not going to solve the congestion problem in the RPU bands alone. While there are many markets that experience crowding on the band, it is not from too many legitimate users but from blatant misuse. MISUSE? Yes, misuse. With exceptions of the TSL channels and the limited area and shared channels, the rest of the band, 161 and 450/455 is to be used in the following order of priority:

1. **Communications during an emergency or pending emergency directly related to the safety of life and property.**
2. **Program material to be broadcast.**
3. **Cues, orders, and other related communications immediately necessary to the accomplishment of a broadcast.**

4. **Operational communications.**
5. **Tests or drills to check the performance of stand-by or emergency circuits.**

In other words an emergency or pending emergency directly related to the safety of life and property has priority over program material. This was demonstrated during hurricane Andrew when the radio and TV stations cooperated to provide emergency communications using their auxiliary links when police, sheriff and ambulance radio systems were unoperable.

However #3 &4 are the ones causing problems with #2, **program material to be broadcast.** Many broadcasters have had revenue producing remotes interfered with by someone using the frequency to dispatch an ENG truck, order dinner or just plain chit chat.

There are many users of the RPU bands that have outgrown not only the use but intent for which the band was set up. For instance a television station that has a 100 watt transmitter on 161.70 at 900' above average terrain used to dispatch an ENG truck makes that frequency unusable for a 100 mile radius of the tower. Most ENG trucks and news cars have simple two way radios that could and should be converted to business band. If a broadcaster 70 miles from the TV tower wants to use 161.70 for a remote broadcast, they can be blown off the air anytime of the day or night by the TV station. 99% of the use by television stations is NOT for broadcast but just for communications.

We need to address the problem of misuse if we are to get more use form the auxiliary broadcast bands. The channels need to be properly prioritized with general communications moved from the auxiliary band to the business band or cellular.

Here is the proposal to prioritize the use as we think it should be:

Proposed 74.403b

1. **Communications during an emergency or pending emergency directly related to the safety of life and property.**
2. **Program material to be broadcast.**
3. **Cues, orders, and other related communications immediately necessary to the accomplishment of program material broadcast on the Part 74.402 frequencies (161, 166, 170, 450/455 MHz auxiliary broadcast frequencies).**
4. **Operational communications when related to program material broadcast on Part 74.402 frequencies (161, 166, 170, 450/455 Mhz. auxiliary broadcast frequencies).**
5. **Tests or drills to check the performance of stand-by or emergency circuits.**

In conclusion I feel that a reallocation of the RPU frequencies will result in more usable frequencies for all broadcasters and that modifing the priority of use table will result in less interference from non-broadcast use.

Illustration #1

ORIGINAL AUXILIARY BANDS
(37 channels)

1606, 1622 and 1646KHz..	**10KHz. bandwidth** **Emission A3E**
25.87 25.91 25.95 25.99 26.03	**40KHz. bandwidth**
26.07 26.09 26.11 26.13 26.15 26.17 26.21 26.23 26.25 **26.27 26.29 26.31 26.3326.35 26.37 26.39 26.41 26.43 26.45 26.47**	**25KHz. bandwidth**
152.87 152.93 152.99 153.05 153.11 153.17 153.23 153.29153.35	**30KHz. bandwidth** **Shared use with industrial users limited use area.**

Illustration #2

AUXILIARY BAND PLAN IN THE MID 1950s.
(60 channels)

(20 universally usable voice channels)

1606, 1622 and 1646KHz..	**10KHz. bandwidth Emission A3E**
25.87 25.91 25.95 25.99 26.03	**40KHz. bandwidth**
26.07 26.09 26.11 26.13 26.15 26.17 26.2126.23 26.25 26.27 26.29 26.31 26.33 26.35 26.37 26.39 26.41 26.43 26.45 26.47	**25KHz. bandwidth**
152.87 152.93 152.99 153.05 153.11 153.17 153.23 153.29 153.35	**30KHz. bandwidth Shared use with industrial users limited use area.**
161.64 161.67 161.70 161.73 161.76	**30KHz. bandwidth**
166.25 170.15	**25KHz. bandwidth Limited use area**
450.15 450.25 450.35 450.45 450.55 450.65450.75 450.85 455.15 455.25 455.35 455.45455.55 455.65 455.75 455.85	**100KHz bandwidth**

Illustration #3

REVISED BAND PLAN 1975
(100 channels)

(53 universally usable voice channels)

1606, 1622 and 1646KHz..	**10KHz. bandwidth Emission A3E**
25.87 25.91 25.95 25.99 26.03	**40KHz. bandwidth**
26.07 26.09 26.11 26.13 26.15 26.17 26.2126.23 26.25 26.27 26.29 26.31 26.33 26.35 26.37 26.39 26.41 26.43 26.45 26.47	**25KHz. bandwidth**
152.87 152.93 152.99 153.05 153.11 153.17 153.23 153.29 153.35	**30KHz. bandwidth Shared use with industrial users limited use area.**
161.64 161.67 161.70 161.73 161.76	**30KHz. bandwidth**
166.25 170.15	**25KHz. bandwidth Limited use area**
450.05 450.15 450.25 450.35 450.45 450.55 455.05 455.15 455.25 455.35 455.45 455.55	**50KHz. channels Communication, program, cues**
450.0875 450.1125 450.1875 450.2125 450.2875450.3125 450.4125 450.4875 450.5125 450.5875450.6125 455.1125 455.1875 455.2125 455.2875455.3125 455.4125 455.4875 455.5125 455.5875 455.6125	**25KHz. channels Communication, program, cues**
450.65 450.70 450.75 450.80 450.85 455.65455.70 455.75 455.80 455.85	**50KHz.. channels Program, cues**
450.925 455.925	**100KHz.. channels Program, cues**
450.01 450.02 450.98 450.99 455.01 455.02 455.98 455.99	**10KHz.. channels Tone signaling, operational com munication, TSL**

Illustration #4

VHF segments:

161.6275, 161.6325, 161.6375, 161.6425, 161.6475, 161.6525, 161 6575, 161.6625, 161.6675, 161.6725, 161.6775, 161.6825, 161.6875, 161.6925, 161.6975, 161.7025, 161.7075, 161.7125, 161.7175, 161.7225, 161.7275, 161.7325, 161.7375, 161.7425,161.7475, 161.7525, 161.7575, 161.7625, 161.7675,161.7725

5KHz segments user picks 6 to make a 30KHz channel

UHF segments:

450.0275, 450.0325, 450.0375, 450.0425, 450.0475, 450.0525, 450.0575, 450.0625, 450.0675, 450.0725, 450.0775, 450.0825, 450.0875, 450.0925, 450.0975, 450.1025, 450.1075, 450.1125, 450.1175, 450.1225, 450.1275, 450.1325, 450.1375, 450.1425, 450.1475, 450.1525, 450.1575, 450.1625, 450.1675, 450.1725, 450.1775, 450.1825, 450.1875, 450.1925, 450.1975, 450.2025, 450.2075, 450.2125, 450.2175, 450.2225, 450.2275, 450.2325, 450.2375, 450.2425, 450.2475, 450.2525, 450.2575, 450.2625, 450.2675, 450.2725, 450.2775, 450.2825, 450.2875, 450.2925, 450.2975, 450.3025, 450.3075, 450.3125, 450.3175, 450.3225, 450.3275, 450.3325, 450.3375, 450.3425, 450.3475, 450.3525, 450.3575, 450.3625, 450.3675, 450.3725, 450.3775, 450.3825, 450.3875, 450.3925, 450.3975, 450.4025, 450.4075, 450.4125, 450.4175, 450.4225, 450.4275, 450.4325, 450.4375, 450.4425, 450.4475, 450.4525, 450.4575, 450.4625, 450.4675, 450.4725, 450.4775, 450.4825, 450.4875, 450.4925, 450.4975, 450.5025, 450.5075, 450.5125, 450.5175, 450.5225, 450.5275, 450.5325, 450.5375, 450.5425, 450.5475, 450.5525, 450.5575, 450.5625, 450.5675, 450.5725, 450.5775, 450.5825, 450.5875, 450.5925, 450.5975, 450.6025, 450.6075, 450.6125, 455.0275, 455.0325, 455.0375, 455.0425, 455.0475, 455.0525, 455.0575, 455.0625, 455.0675, 455.0725, 455.0775, 455.0825, 455.0875, 455.0925, 455.0975, 455.1025, 455.1075, 455.1125, 455.1175, 455.1225, 455.1275, 455.1325, 455.1375, 455.1425, 455.1475, 455.1525, 455.1575, 455.1625, 455.1675, 455.1725, 455.1775, 455.1825, 455.1875, 455.1925, 455.1975, 455.2025, 455.2075, 455.2125, 455.2175, 455.2225, 455.2275, 455.2325,455.2375, 455.2425, 455.2475, 455.2525, 455.2575, 455.2625, 455.2675, 455.2725, 455.2775, 455.2825, 455.2875, 455.2925, 455.2975, 455.3025, 455.3075, 455.3125, 455.3175, 455.3225, 455.3275, 455.3325, 455.3375, 455.3425, 455.3475, 455.3525, 455.3575, 455.3625, 455.3675, 455.3725, 455.3775, 455.3825, 455.3875, 455.3925, 455.3975, 455.4025, 455.4075, 455.4125, 455.4175, 455.4225, 455.4275, 455.4325, 455.4375, 455.4425, 455.4475, 455.4525, 455.4575, 455.4625, 455.4675, 455.4725, 455.4775, 455.4825, 455.4875, 455.4925, 455.4975, 455.5025, 455.5075, 455.5125, 455.5175, 455.5225, 455.5275, 455.5325, 455.5375, 455.5425, 455.5475, 455.5525, 455.5575, 455.5625, 455.5675, 455.5725, 455.5775, 455.5825, 455.5875, 455.5925, 455.5975, 455.6025, 455.6075, 455.6125.

5KHz segments user picks 5 for 25KHz channel 10 segments may be "stacked" for a 50 KHz. channel

450.6375, 450.6625, 450.6875, 450.7125, 450.7375, 450.7625, 450.7875, 450.8125, 450.8375, 455.6375, 455.6625, 455.6875, 455.7125, 455.7375, 455.7625, 455.7875, 455.8125, 455.8375,

25KHz. segments user picks two for 50KHz. channel

Illustration #5

PROBLEMS WITH 1985 BAND PLAN
Channel offset from standard band plan by 5KHz.

161.6275, 161.6325, 161.6375, 161.6425, 161.6475, 161.6525
Existing standard frequency 161.64

161.6325, 161.6375, 161.6425, 161.6475, 161.6525, 161.6575
Center frequency 161.645

Channel offset from standard band plan by 15KHz.

161.6275, 161.6325, 161.6375, 161.6425, 161.6475, 161.6525
Existing standard frequency 161.64

161.6425, 161.6475, 161.6525, 161.6575, 161.6625, 161.6675
Center frequency 161.6550

161.6575, 161.6625, 161.6675, 161.6725, 161.6775, 161.6825
Existing center frequency 161.67

Illustration #6

PROPOSED NEW ALLOCATION, 1993
(124 channels)

(85 universally usable voice channels)

1606, 1622 and 1646KHz..	**10KHz. bandwidth Emission A3E**
25.87 25.91 25.95 25.99 26.03	**40KHz. bandwidth**
26.07 26.09 26.11 26.13 26.15 26.17 26.21 26.23 26.25 26.27 26.29 26.31 26.33 26.35 26.3726.39 26.41 26.4326.45 26.47	**25KHz. bandwidth**
152.87 152.93 152.99 153.05 153.11 153.17 153.23 153.29 153.35	**30KHz. bandwidth Shared use with industrial users limited use area**
161.6275 161.6526 161.6775 161.7025 161.7275 161.7525 161.7725	**25KHz. bandwidth**
166.25 170.15	**25KHz. bandwidth Limited use area**
450.050 450.075 450.100 450.125 450.150 450.175 450.200 450.225 450.250 450.275 450.300 450.325 450.350 450.375 450.400 450.425 450.450 450.475 450.500 450.525 450.550 450.575 450.600 450.625 450.650 450.675 450.700 450.725 450.800 450.825 450.850 450.875 450.900 450.925 450.950 455.050 455.075 455.100 455.125 455.150 455.175 455.200 455.225 455.250 455.275 455.300 455.325 455.350 455.375 455.400 455.425 455.450 455.475 455.500 455.525 455.550 455.575 455.600 455.625 455.650 455.675 455.700 455.725 455.800 455.825 455.850 455.875 455.900 455.925 455.950	**25KHz. bandwidth subject of special division by base, mobile stations and fixed and mobile repeaters.**
450.01 450.02 450.980 450.990 455.010 455.020 455.980 455.990	**10KHz bandwidth TSL only**

USING ISDN, T1 AND SWITCHED 56

Thursday, April 22, 1993

Moderator:
Jan Kowalcyzk, KYW-AM, Philadelphia, Pennsylvania

ACCESSING THE PUBLIC SWITCHED DIGITAL NETWORK
James M. Switzer
RE America, Inc.
Westlake, Ohio

PANEL
Panelists:

Larry Hinderks
Corporate Computer Systems
Holmdel, New Jersey

Angela DePascale
Global Digital Datacom Services, Inc.
Farmingdale, New York

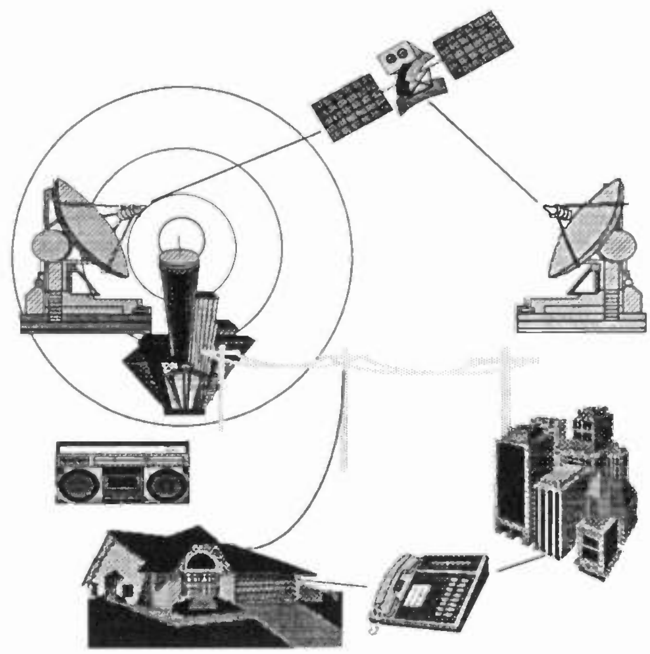
Tom Ballister
Controlware, Inc.
Holmdel, New Jersey

ACCESSING THE PUBLIC SWITCHED DIGITAL NETWORK

James M. Switzer
RE America, Inc.
Westlake, Ohio

ABSTRACT

Digital audio is a buzzword that has captured the attention of the broadcast industry for several years. Digital audio for broadcasters represents, among other things, parity with the high audio quality of Compact Disc technology that the consumer has come to expect. Aside from understanding the technology behind linear or compressed digital audio, the transportation of digital audio across the public telephone networks has added new requirements in the educational process involved in broadcast engineering. As a manufacturer of digital audio codecs, RE America finds it helpful to provide a guide to broadcasters for accessing the digital pipeline.



INTRODUCTION

Switched digital services are rapidly growing in popularity throughout the world. The greatest use of switched digital transport technology was intended for transmission of computer data at rates significantly greater than available through common PC modems. This saves time and money for file transfer applications and on-line systems with remote terminals.

Business use of switched digital services has focused on high compression video for teleconferencing, allowing business meetings to take place between remote locations without the expense of employee travel. Many other uses of the switched digital networks save time and money using high data transmission rates.

In the past few years, the broadcasting industry has been using switched digital services for terrestrial transmission of digital audio information.

As a manufacturer of a wide variety of digital audio codecs for broadcast use, RE America recognized that our customers would benefit from a reference guide for accessing the public switched digital networks. There are many new standards, terms and infamous telco acronyms to adhere to and understand. RE America offers this guide as a basis for familiarization with these new telephone services. Any updates and comments from broadcasters are appreciated. Portions of the RE guide are incorporated in this paper.

AN OVERVIEW OF SWITCHED SERVICES

Various areas of the country offer different types of digital service. Switched 56 service is currently the most widely available "cost effective" service, providing 56 kbps transmission speeds which can be used to transport information across town or around the world. Switched 56 service is a precursor to ISDN (Integrated Services Digital network), and grew in the U.S. out of demand for digital services by customers who could no longer wait for ISDN availability. Prior to the availability of switched 56 service, customers had to use higher cost dedicated services such as 64 kbps, T-1 and T-3.

ISDN has been an industry buzzword for two decades, but lack of standardization agreements has slowed its growth. With the accepted National ISDN-1 (N-ISDN-1) standard in place, the United States will now begin to benefit from widespread ISDN deployment. Even with this standard there will be a gap in total inter-operability for the next few years as some phone companies will lag in the deployment of the necessary signaling systems required to bring N-ISDN-1 to every point of presence in the U.S. Early ISDN software versions may not accept the common signaling system defined by N-ISDN-1. In general, however, N-ISDN-1 will be available in most major cities by the end of 1993. By the end of 1994, N-ISDN-1 is expected to be available to between 40% and 50% of the U.S. market and fully deployed by 1997. Within this time frame, the standard may evolve to proposed versions N-ISDN-2; and possibly, N-ISDN-3.

DESCRIPTIONS OF SWITCHED SERVICES

POTS, Plain Old Telephone Service

POTS is traditional telephone service as most people know it. It is truly universal in that virtually anyone can directly dial any other telephone in the world. Unfortunately, it is highly bandwidth limited, generally between 300 and 3300 to 3400 Hz. While POTS supports data transmission, it too is limited, generally to up to a maximum 19,200 baud with sophisticated modems, but typically only up to 9600 baud. Data transmission on POTS sends digital information through an analog modulation process which works within the bandwidth of the typical phone network.

The inherent limitations of POTS service has generated the demand for higher speed digital services. While POTS is limited in provided the capabilities of advanced digital services, it is the model for the future global switched network, where high speed digital service can be dialed from anywhere to anywhere. This concept is more commonly associated with ISDN (Integrated Services Digital Network).

Dry Copper

Dry Copper, (DC pairs) has been a staple telephone company provided service to the broadcasting community since the earliest days of radio broadcasting. With the advent of digital telephone technology and the growth and dedication to fiber optic electronics most Local Exchange Carrier (LEC) interoffice trunking is now done digitally, either through fiber optic cable or dedicated T-1 copper links. This technology growth has limited the present availability of DC pairs generally to within the boundaries of a single central office serving area.

Many broadcasters still use DC pairs for remote control capabilities in the U.S. As a supplier, RE America recognizes the value of available DC pairs and addressed the needs of many broadcasters who can utilize these often less expensive circuits, for high quality digital audio links. The RE 8720/8730 15 kHz Stereo Tie Line, or the RE 8721/8731 Mono Tie Line digital audio codec takes advantage of the availability of DC pairs by offering the broadcaster a digital means of bypassing the traditional telco equalized line services. This allows the broadcaster to regain control of the equalization process on the critical STL path. The RE Tie Line codecs offers a transparent means of connecting the studio to the transmitter with virtually no group delay, line induced noise or roll-off in the frequency response which is inherent in using non-equalized DC pairs or even most phone company provided equalized lines.

Dry Copper Continued...

DC pairs are generally a fraction of the cost of telephone company equalized line charges. However, some telephone companies tariff their DC pairs by usage. In various areas of the U.S., the LEC may charge equalized line costs for DC pairs if they are used for audio transmission.

Switched 56 Service

Customer demand for switched digital services at higher speeds grew faster than the ISDN standardization and implementation process could meet. A stopgap measure at 56 kbps was developed and implemented with enough standardization to service these needs. Switched 56 digital services are generally available throughout the U.S. Early implementation of some form of ISDN has allowed for limited availability of switched 64 kbps services as well.

Switched 56 services is provided under a variety of service names by the LECs and Inter Exchange Carriers (IXC). To the general business world, switched 56 services represent a means of transporting data at a rate at least five time faster than that of POTS service. This allows for a significant cost savings for large data users who need to share computer files outside of their respective office environment, especially when more than one outside location requires data communications. From a business standpoint, there are a variety of scenarios for cost comparisons based on actual user needs. In some cases, dedicated point to point digital service, which is generally more expensive than switched 56, is more cost effective between two points which require several hours of continuous connection time.

This is not necessarily the same case for broadcasters who are looking to use the high speed switched public networks for program audio transportation. The advantage of transporting high quality audio over any distance may include advantages that outweigh the typical business cost comparison calculations.

While switched 56 service have obtained a solid foothold through customer demand, and promises to be available for several years to come, they are still considered the pre-cursor to ISDN services. As ISDN availability grows over the next few years, switched 56 usage will decrease, mainly because ISDN will cost less to install and use while offering more capability. Presently the cost of switched 56 service ranges from near equal to POTS to upwards of 10 times its cost for monthly recurring charges.

Dedicated Digital Service

Dedicated Digital Service (DDS) is a point to point only digital service, generally available at 56, 64 and T-1 digital transmission rates. While usually more expensive than the switched services, it is more widely available. Dedicated digital services are most cost effective for continuous connections under constant usage loads. It can only be used from one point of termination to another, however; it may be the only way to obtain digital service from your location to the LEC central office which houses the IXC's Point Of Presence (POP).

ISDN

Integrated Services Digital Networks are being implemented worldwide. Many European countries have enjoyed the use of ISDN for several years now. Some countries, such as France, have 100 % deployment of the ISDN. U.S. deployment has been delayed by standards processes inherent in meeting the needs of a country the size of the United States, where the "public" networks involve many public and private companies, cooperating to build a transparent common network. Growth and modernization is spurred by customer demand and available capital. The nature of a major "utility" such as the telecommunications infrastructure of the U.S. is subject to tight restrictions by government regulation because of its inherent "monopolistic" operation.

After many years of planning, standardization, tariff definitions and general legislative work, the U.S. network providers are now deploying ISDN service. While ISDN has been defined for service rates into the billions of bits per second (Giga bits per second),

ISDN Continued...

the most common public awareness of ISDN is at the BRI (Basic Rate Interface) and PRI (Primary Rate Interface) levels. BRI service includes two 64 kbps B-channels for circuit switched voice or data traffic, and one 16 kbps D-channel for signaling and potential additional data up to 9600 baud. PRI is similar to non-ISDN T-1 transmission, but allows for more flexible features in defining the use of its 23-B channels and one D-channel.

ISDN service is designed to operated in the same way as POTS. In other words, with an ISDN, the user can dial from anywhere on an ISDN circuit, to another ISDN circuit. This is sometimes referred to as the "global ubiquitous network".

T-1 Service

T-1 service refers to customer controlled equipment which can access up to 24 DS0 (24 X 64 kbps) circuits for a total of 1.536 Mbps. The full service equals 1.544 Mbps and includes 8 kbps for signaling. With T-1 service, a large amount of data can be transported point to point, or several DS0 "slots" can be user configured to disseminate or receive to or from several points. T-1 service can be used as "fractional T-1" or part of a full T-1.

Pricing for T-1 service tends to be high ($600 to $1,400 per month). Fractional T-1 service usually requires a full T-1 access line (and charge) to the local CO. T-1 and fractional T-1 is most commonly used by broadcasters for "linear" non-compressed digital audio, or data rates higher than that of 56 kbps, 64 kbps or two times these services. Broadcasters with high quality audio requirements needing long term or continuous service such as program distribution or even STL applications will benefit most from T-1 terrestrial services.

ORDERING A SWITCHED DIGITAL SERVICE

In general, it is best to simply call the business services division of your local telephone service provider when ordering a switched digital service.

You may also call authorized service agents, who may be more helpful in handling your exact needs. You do not have to pay an agent more for his service as it is usually paid by the LEC as an agent commission fee. You can identify local agents in the "Yellow Pages" under "telephone (services)" or "computer (services)".

A chart is available in the full RE America Digital Network Access Guide which describes general pricing guidelines by the Regional Bell Operating Companies as well as other Local Exchange Carriers (LEC) and Inter-Exchange Carriers (IXC). These rates are very general by region, and can change based on distance from the customer site to the serving Central Office. Most important is the fact that the LEC or IXC may be filing updated tariffs which will affect published pricing.

Charges:

For current pricing, always consult your local operator or authorized service agent.

There are three categories of charges you will run in to. First is the "Non-Recurring" charge or installation charge. Second is the "Recurring Charge" or monthly fixed line charge. The third charge is the "Access Charge" or the per minute usage charge. Some IXC charges are based on 6 second intervals as well. Each category can be defined further as follows:

Non-Recurring Charges:
Switched 56: Administration Design and CO Connection Customer Connection
ISDN: CO Termination Circuit Switched Data (2B) Service Establishment Charge

Charges Continued...

Recurring Charges:
Switched 56:
Local Distribution Channel
Channel Mileage Termination
Channel Mileage
CO Bridging (Multipoint Circuits only)
(Other "surcharges" may apply)
ISDN:
Line charge
CO Termination Charge
Circuit Switched Data (2B)

Access Charges
Switched 56:
First Minute
Additional Minutes
ISDN:
Measured Service Day/Night
Rate Mileage
Initial Minute
Additional minutes

Each category in the previous chart may or may not apply in every case and certainly varies from region to region, and from one provider to another. The Non-Recurring charges for installation, the Recurring monthly charges and if applicable, the Access fees are charged by the LEC or local service provider. The IXC's in most cases only charge for Access, just like a POTS long distance service.

When using an Inter Exchange Carrier such as AT&T, MCI, Sprint or WilTel, expect to pay for all of the charges to the Local Exchange Carrier listed above as well as the additional access charges by the IXC. If connected to the IXC via a dedicated digital service, there will not be any local access charges from the Local Exchange Carrier. If using an existing T-1 service, designating a DSO "slot" for switched 56 service to the Point Of Presence of the Inter-Exchange Carrier will by-pass the Local Exchange Carrier charges.

CSU's/DSU's AND TERMINAL ADAPTERS

In order to connect digital audio codecs to the public switched digital networks a terminal adapter which can perform the necessary signaling protocols for dial up capability and network compatibility must be utilized. Switched networks use either two wire or four wire connections which, while different in protocol, are able to communicate between a variety of manufacturer's equipment.

CSU's, DSU's and terminal adapters perform dialing functions, re-dial, number storage and a whole host of other functions. They come with or without key pad dialers (some can pre-store numbers and then scroll a display for autodial use) and many can be operated from a PC terminal if desired.

Terminal costs can run from as low as $595 up to $2,000 for typical models which serve the switched 56, dedicated 56/64, switched 384 and ISDN applications. T-1 terminal adapters can be higher in cost depending on the required functions.

Many broadcast facilities are equipped with PBX phone systems, or use Centrex remote PBX services. PBX manufacturers may have add on boards to accommodate ISDN service. It is best to consult with the phone system provider for these options.

In most cases, the local exchange carrier or authorized service agent can recommend or supply the required interface for the digital service needed. Many full service agents can provide sale or lease packages for CSU's, DSU's and terminal adapters. Two full turnkey agents are listed at the end of this article for reference purposes.

RE America, will introduce a MUSICAM ISO/MPEG Layer II/IIA digital audio codec with a built in ISDN terminal adapter. The terminal adapter will be able to reverse multiplex up to six 64 kbps B-channels for a total of 384 kbps transmission. This saves the user the cost of up to three ISDN terminal adapters which can prove to be extra baggage for remote broadcast usage.

GLOSSARY OF TELECOM TERMS

The following is a list of the most commonly used terms associated with digital telephone network services.

General Network Terminology

Bridging - The ability to connect one point to many. Usually done within a LEC or IXC CO.

Central Office (CO) - A switching center facility for common carriers. In the local exchange telephone company, a CO is the switching center for calls to the 10,000 group exchange numbers associated with a unique area code and three digit prefix. A typical LEC CO houses from 3 to 10 prefixes.

Centrex - Telephone service that offers PBX type service to individual lines within a business customer environment such as hold, conference, extensions, transfer and other typical business phone features without housing the switching equipment on the customer premise.

CSU - Customer Service Unit, Channel Service Unit; a data terminal which interfaces digital data at rates usually up to 56 kbps to the digital network. The CSU handles the technical standards for loopback testing, signaling (dialing) and other requirements which meet FCC part 68 for connecting equipment to the telephone networks.

DAL - Dedicated Access Line; a non-switched access line that connects an inter-LATA or inter-state user to an IXC (inter-exchange carrier, long distance carrier) through the LEC (Local Exchange Carrier).

DDS - Digital Data Service; a dedicated digital data transmission service that typically offers transmission speeds up to 64 kbps over interconnected point to point digital private lines.

four wire - A two-way transmission circuit utilizing separate paths for transmit and receive. Four wire digital network services generally allow 18,000 cable feet between the customer termination and the local CO when using 26 gauge wire. Four wire switched 56 service may be offered from the following CO switches: 1A, 4ESS, 5ESS and DMS-100.

LATA - Local Access and Transport Area; an area served by the LEC. Intra-LATA is within the LATA, inter-LATA is between two or more LATA's.

LEC - Local Exchange Carrier; the telephone service provider within a given LATA.

Dry Copper - An unloaded copper telephone line with dedicated customer termination points. Also referred to as Metallic Pairs or DC pairs, dry copper does not pass through switching equipment at the CO. Dry copper is usually available only through one LEC CO boundary.

DS0 - Digital Signal Level "Zero"; The baseline for digital hierarchy equal to one 64 kbps pulse code modulated transmission channel.

DS1 - Digital Signal Level "One"; 1.544 Mbps transmission based on the multiplexing of 24 DS0 channels plus 8 kbps for signal control. Also referred to as **T-1**.

DS2 - Digital Signal Level "Two"; 6.312 Mbps transmission based on a grouping of four multiplexed DS1 channels plus 156 kbps for signal control. Generally uncommon in availability for commercial use but may see growth with MPEG video compression usage.

DS3 - Digital Signal Level "Three"; 44.6 Mbps transmission based on a grouping of 28 multiplexed DS1 channels plus 1.368 Mbps for signal control. Also know as **T-3**.

DSU - Data Service Unit, Digital Service Unit; a data terminal which interfaces digital data at rates usually from 64 kbps and up to the digital network. The DSU handles the technical standards for loopback testing, signaling (dialing) and other requirements which meet FCC part 68 for connecting equipment to the telephone networks.

General Network Terminology Continued...

IXC - Inter-Exchange Carrier; sometimes noted as "IEC", the IXC is a long distance service provider such as AT&T, MCI and Sprint.

kbps - Kilo Bits Per Second; one thousand bits per second times the number preceding it.

Mbps -Mega Bits Per Second; one million bits per second times the number preceding it.

NIU - Network Interface Unit; the unit between a codec and the digital network that provides the necessary interface required for digital transmission , including dialing capabilities. NIU's include CSU, DSU, Terminal Adapters (TA) or Terminal Interface Units (TIU).

POTS - Plain Old Telephone Service; traditional residential and business telephone service, generally limited to between 300 and 3400 Hz bandwidth.

POP - Point Of Presence; the Central Office location where the Local Exchange Carrier interfaces to an Inter-Exchange Carrier.

reverse multiplexing - The process of synchronizing two data circuits for additional transmission speeds by the customer, rather than the phone network. An example is two times switched 56 for 112 kbps speeds, or in the ISDN, 2B, reverse multiplexed for a total of 128 kbps transmission speed.

Tariff - A filing by the Local Exchange Carrier to the FCC for approval of proposed service charges and access rates based on capital plant and equipment, operating expenses and profit margins desired as associated with the LEC serving area.

tie line - A leased or private dedicated telephone circuit provided by common carriers that links two points together without using the switched network. Usually available within the boundaries of a central office serving area (see Dry Copper).

two-wire - A two-way transmission circuit utilizing the same pair for transmit and receive. Two wire digital network services allow for lower transmission rates for bandwidth on demand on switched 56 service but is generally limited to customer termination connectivity within 13,100 cable feet from a CO. Two wire switched 56 service indicates that the LEC is using a "1A Switch" or a "DMS-100 switch" in the CO. ISDN utilizes two wire technology.

ISDN SPECIFIC TERMS

B channel - Bearer channel; a single ISDN 64 kbps digital channel.

broadband ISDN - While still being defined in terms of standardization, the U.S. market generally refers to any ISDN service at or exceeding 1.544 Mbps as being broadband. Allocation of required data rates are defined by the end user. Formal international definitions define broadband ISDN as 150 Mbps and higher, subdivided by users per their specific needs. Future provisions allow in excess of 1 billion bits per second (1 giga bits per second).

BRI - Basic Rate Interface; single line ISDN service (two wire) standards provide for two B-channels and one D-channel allowing two times 64 kbps (128 kbps) plus 16 kbps for signaling. Upon dialing and completing a connection, the D-channel can be used for additional user data requirements up to 9600 baud (packet switched). The total BRI equals 144 kbps.

D Channel - Signaling and data channel for ISDN service. The D-channel is a packet switched path for sending network control signals that pass dialing information and connection confirmation, caller ID and other network information. The D-channel uses 16 kbps of bandwidth for BRI and 64 kbps of bandwidth for PRI ISDN service.

narrowband ISDN - Refers to ISDN service at single B-channel through 23 B-channel (PRI) transmission rates.

ISDN SPECIFIC TERMS Continued...

network interface -Point of demarcation where the telephone company responsibility for maintenance ends and consumer equipment connections begin.

N-ISDN-1 - National Integrated Services Digital Network 1; an ISDN standard intended to allow all ISDN providers to follow a common signaling system approach for ubiquitous transparent inter-connectivity between service providers.

NT-1 - Network Termination - 1; an active device that connects the customer premise equipment to the LEC point of demarcation (U interface).

PRI - Primary Rate Interface; interface standard for T-1 ISDN service at 1.544 Mbps (North America), comprised of 23 B-channels and one D-channel.

R interface - The point where a non-ISDN codec, telephone or data terminal connects to an ISDN terminal adapter.

S interface -The point where an ISDN terminal adapter or ISDN compatible codec or other data device connects to an ISDN compatible PBX, local area network or other ISDN compatible multistation system.

T interface - The point where an ISDN compatible PBX, local area network or any other ISDN compatible mutistation system connects to ISDN channel termination equipment. In a single line (BRI), the T interface is the same as the S interface.

Terminal Adapter - Similar to CSU or DSU, the terminal adapter handles the dialing and other switching features on the ISDN.

U interface - In the U.S., the U interface serves as the network interface point of demarcation between the public switched network (LEC/IXC) and the customer premise location.

CONCLUSION

As more broadcasters place digital audio codecs in to service it is ever more important to educate the broadcast industry in the procedures required to obtain and use the public switched digital networks. RE America offers a guide to this process which includes current average service costs for digital service. As a service to the broadcaster who is entering this new dimension of broadcasting, RE America welcomes requests for the guide. Please call or write to:

RE America, Inc.
31029 Center Ridge Road
Westlake, Ohio 44145
Phone:(216)-871-7617 FAX(216)-871-4303

ACKNOWLEDGMENTS

Information for this article has been obtained from the RE America, Inc. Digital Network Access Guide. The guide is available from RE at the above address or can be obtained from any one of the RE distributors. Call for distributor information.

ADDITIONAL SOURCES OF INFORMATION

Many agents and representatives sell digital services as a turnkey package. Agents can be found in local telephone listings. Two national agents are BUSINESS LINK at 1-800-929-LINK or Global Digital Datacom Services Inc. at (516) 694-6805. RE America, Inc. is not associated with, nor endorses the service of any one agency for your needs, but will endeavor to make information available and current through the RE Digital Network Access Guide..

THE IMPACT OF CONSUMER GRADE EQUIPMENT ON TELEVISION BROADCASTING*

Thursday, April 22, 1993

Moderator:
J. Robinson, Hearst Broadcasting, Milwaukee, Wisconsin

*Papers for this session were not available at the time of publication.

CAMERA WORKSHOP*

Thursday, April 22, 1993

Moderator:
Kelly Williams, NAB, Washington, District of Columbia

A hands-on demonstration of the setup and maintenance of modern video cameras.

*Papers for this session were not available at the time of publication.

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