PROCEEDINGS

31ST ANNUAL BROADCAST ENGINEERING CONFERENCE

NATIONAL ASSOCIATION OF BROADCASTERS

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NATIONAL ASSOCIATION OF BROADCASTERS
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Landmark Communications
Norfolk, Va.

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Welcoming Remarks

Vincent T. Wasilewski
NAB President

Good morning and welcome to the 31st Annual NAB Broadcast Engineering Conference. For over 30 years now this industry has demonstrated remarkable progress in the field of electronics.

One of the purposes of this Convention every year is to allow us to meet and hear reports and updates on the new breakthroughs that have helped us to achieve such a high standard of excellence in broadcasting.

Just a short tour of the many exhibits will give you some idea of the contributions and the ingenuity of our engineers and technicians. Perhaps we take this all for granted, but we should always remember that it is technology which is the basic component of our broadcasting industry. It IS broadcasting.

This conference provides the opportunity to pick up on the latest developments and discuss the various techniques used by your respective stations. There are over 12,000 people involved in the NAB Convention; over 89,000 square feet of exhibits. It is the first time ever that Washington's three largest hotels have had all their exhibit space committed!

Throughout the year I am asked by engineers just what our Vice President for Engineering, George Bartlett, actually does. In one sentence . . . George is instrumental in helping the NAB convince the FCC to simplify all those regulatory restrictions that make your jobs more and more difficult. When Automatic Transmission Systems become a way of life at every station in the country, George Bartlett, to a great degree, will be the person who led the parade.

Other significant areas of concern in which our Engineering Department has been active this year, areas of concern to each and every one of you, are:

- Citizens Band interference to the reception of AM/FM/TV broadcast service.
- Restructuring the non-commercial portion of the FM band to prevent interference to TV channel 6.
- The World Administrative Radio Conferences for 1977 and 1979.
- Restructuring of the Rules and Regulations in the Auxiliary Broadcast Service.
- The implementation of Automatic Transmission Systems.
- AM Stereo.
- and in addition the completion of the final section of the NAB Engineering Handbook.

Needless to say, its been a very busy year.

Let me extend my congratulations to Dan Smith, retired Senior Vice President for Engineering at Capital Cities Communications, for being chosen to receive this year's Engineering Achievement Award. Dan has exemplified the best in broadcast engineering and his contributions to the field have been tremendous. He is a fine, dedicated man whose professional life has been geared entirely to the cause of better broadcasting.

I hope that your purpose in being here in Washington at this Conference is Discovery. I hope that you are here to learn, to seek information, and to exchange ideas. It is this kind of curiosity, dedication and willingness to experiment with untried formulas that makes American broadcast engineering the best in the world.

Washington is a wonderful place to be. Right now the town is full of new blood, new ideas and new hopes. Enjoy your visit and have a terrific Convention.

MAN OF THE YEAR....

Daniel H. Smith Wins Engineering Award

Mr. Smith (left), retired senior vice president for engineering, Capital Cities Communications, receives NAB's 1977 Engineering Award from George W. Bartlett, NAB vice president for engineering, at Tuesday's luncheon. The citation is at right.

NATIONAL ASSOCIATION OF BROADCASTERS

ENGINEERING AWARD

Presented to

DANIEL H. SMITH

In recognition of his distinguished professional career ...

For his many contributions to our Nation's knowledge in the field of communication technology. . .

For his untiring efforts to foster advances in the art of broadcasting . . .

And for his pioneering spirit which has so richly enhanced the forward progress of broadcast engineering

31ST ANNUAL
BROADCAST ENGINEERING CONFERENCE

March 29, 1977 Washington, D.C.

What's New at the Exhibits

Doyle D. Thompson
Chairman,
Broadcast Engineering Conference Committee
Director of Broadcast Engineering
Landmark Communications
Norfolk, Va.

Last December when the Conference Committee was planning the engineering program, we discussed the problems that would eventually crop up with the three hotel exhibit concept.

I don't need to remind you of the multiplicity of logistic problems, the least of which was assigning space in the various hotels to an overflow of exhibitors, all of whom seem to have some degree of vested rights based upon size and longevity of association with the NAB.

After that problem had been resolved, the burning question was "How do we move 7000 conferees between the various exhibit halls?"

And lastly "How do we make sure that the conferees see all that brand new gear without missing anything?"

The solution to the space assignment problem was relatively easy. We just delegated that responsibility to our ingenious exhibit manager, Ed Gayou, who I'm sure you have seen scurrying around the exhibit area in a very official looking manner.

The problem of moving people between the respective exhibit locations was easily solved by calling METRO, the Metropolitan Transit Agency, which quickly suggested a shuttle bus service which the NAB eagerly subscribed to.

The last, but not least, problem was how to pass-the-word to all conferees as to what equipment is where so that when we all return home we don't meet a fellow attendee on the street and echo the classic remark: "I didn't see that!" Sometimes I think we should have a post-convention exhibit just to accommodate the many hundreds, if not thousands, of the "I missed that completely" group.

After considerable soul searching, the Committee recommended that I, as Chairman of the Conference Committee, should present this overview of the equipment area. And that, in a nutshell, is why we are all sitting here at the moment.

The first step in the preparation of this portion of the program, was to canvass the several hundred exhibitors who were assigned space in either the Shoreham Americana, the Sheraton Park or the Washington Hilton hotels.

You would think a call to all exhibitors outlining our plan and requesting information on new products would have generated a response that would have overwhelmed us. One would think that any manufacturing organization would be "exposure conscious" and would supply us with material it would require a barge to move.

Unfortunately, this was not the case. Only 35 of the faithful rose to the occasion and responded to the call. Maybe those that didn't need new PR people or something. What a golden opportunity to have missed.

For the purposes of this presentation and to simplify the talk so that we are not hopping from hotel to hotel, I will talk first about those exhibits located in the Shoreham, then move on to the Sheraton Park and lastly discuss those at the Washington Hilton.

Needless to say, this presentation cannot and is not intended to be all-inclusive, due both to the circumstances and time alloted. But hopefully it will focus on some of the new innovations that you will be seeing and hearing about in the next few days.

The Shoreham Americana . . .

This is not only the Engineering Conference hotel but houses a number of exhibits, including public service exhibits that are important to our industry.

The primary public service exhibitor is the Federal Communications Commission. It was a booth in the Birdwalk where FCC staff members are available to discuss specific problems relating to Commission regulations and station operations. In addition, on Wednesday, the FCC monitoring van will be available for inspection on the parking lot just behind the FCC booth. This is your golden opportunity to talk to the Commission's field personnel who actually conduct station inspections.

Turning now to the equipment exhibitors at the Shoreham . . .

Lightning Elimination Associates has a number of new innovations to eliminate this troublesome phenomena. Lightning has been a major cause of equipment losses and operational downtime which can only be coped with through the implementation of a totally isolated communications site such as that now provided by Lightning Elimination Associates. Its display includes a dissipation array for tower protection, a multiple transient eliminator, a main power line surge eliminator, a low power surge eliminator, and a transient eliminator for data lines. All are newly-developed products for the control and taming of lightning strikes.

Berkey Colortran is introducing a lightweight, high intensity HMI daylight light source utilizing completely electronic self-contained ballast. This is a major breakthrough in ballast design that makes this product feasible for Electronic News Gathering and studio applications. Another new product is a very low-cost 100 pre-set memory lighting control system. For the first time, such a system is available at a price less than a 3-scene preset. The memory may be used in the studio or as a completely portable system. It is a "must-see" for those who are concerned with lighting for ENG, and who isn't at this point in time?

The **Winsted Corp.** is exhibiting the first video console that lets you combine all your editing equipment into a single movable unit. This compact, totally modular editing console accommodates all sizes of VTR's, editors, monitors and equipment, yet can be placed virtually anywhere. It can even be rolled into a van to create a mobile unit setup. The console is engineered for easy adaptability and rugged strength and components moved easily and locked safely in place. The VTR rack has a sliding pullout shelf and the editor shelf expands with a pullout work surface. Removable back panels allow full rear access to power cords and cables.

Cinema Products Corp. has on display its revolutionary new Steadicam camera stabilizing system which has gained widespread acceptance not only when used with the RCA TK-76 camera but has now been adapted to accept 16 and 35mm motion picture cameras. The Steadicam greatly increases the creative latitude of the cinematographer and the director by solving the problem of image steadiness when shooting with either a hand-held video or film camera. The system has recently been used in filming such productions as *Bound for Glory, Marathon Man, Rocky* and numerous other epics.

Television Equipment Associates has several new products which will be of interest to television engineers. One is the Matthey Automatic Video Equilizer. With the acceptance of unmanned automatic transmitters, the Matthey Equalizer will give automatic control of 10 Video parameters. In addition to normal video gain, burst phase, etc., the unit controls 2T phase and gain chroma delay, sync, LF bar tilt — all automatically. Another product on display is the Matthey TV line selector. Many TV engineers use an oscilloscope to look at TV line information, but this is tiring and frustrating because of jitter, loss of triggering, etc. Waveform monitors solve the problem but are expensive and not easily available. This new Matthey device just adds to any oscilloscope and gives the user a waveform monitor facility at low cost, just about a must for these old tired eyes . . .

Garner Industries has on display a new line of high-speed tape erasers. The new Garner video eraser can easily erase video cassettes in one 4-second "hands-off" operation. Tapes are placed on a continuous belt and passed through high flux coils to produce an erasure which meets or exceeds professional recording standards.

Gotham Audio Corp. is displaying the new Telefunken Telcom C4 Noise Reduction System. This is a 4-ban companding system that will increase the dynamic range of a transmission or recording system by 30db without noise pumping or the need for alignment of the expander to track the compressor. Its short attack time prevents over-modulation of the recording or transmission system and transfer characteristics are independent of temperature.

IGM, a division of **NTI,** is showing a prototype of its new Magna Carter as well as its previously shown Go-Cart, which has now been expanded to handle 78 cartridges. The new Magna Carta tape storage system can handle up to 1000 cartridges and is automated to remove any given cartridge from its storage cubicle. It will play it as programmed, then return it to storage. This is a stand-alone product made possible through the use of microprocessing technology. This a must-see for those involved or about to become involved with or in automation systems.

BJA Systems is offering a colorizing service for new and old film and tape. This is a service rather than an equipment system. Cost is directly related to the length, quality and purpose of the program material. Specifications required by the customer also affect the price. A rough base is $5000, for a half hour program.

ESE is exhibiting for the first time and offers five new products: 1) a hand-held time calculator; 2) a low cost SMPTE Time Code Reader; 3) a Timer/Source Interface, 4) console mounting slaves; and 5) a Clock/Timer with memory that remembers the time of day while being used as a timer.

Duca-Richardson Corp. is introducing a new and remarkably different video production switching system. Among its features: 1) color coding; 2) 15 button keyboards; 3) 10 key sources; 4) production modules; 5) 99 time choices; and 6) 99 pattern choices and status lights. All you switcher buffs are urged to look at this mind-boggler.

Otuari Corp. introduces a new generation of compact professional recorders with two channels on quarter-inch or four channels on half-inch tape, built-in, DC servo, pitch control, separate transport and electronics and an interface jack for DBX or Dolby Noise Reduction systems. In addition, an automated station reproducer is on display.

Sintronic Corp. has on display its latest 1000 watt solid-state AM transmitter which features a lower number of transistors. The transmitter has a unique circuit which monitors the output and adjusts the drive for a change in load, thus maintaining the output power transistors in their safe operating area. The transmitter is capable of 125% positive modulation into various loads from 15-ohms to open circuit . . .

Time and Frequency Technology Inc. has two new devices on display. One is a modular system for the digital remote control of AM/FM/TV transmitters. This new system is adaptable to ATS operation and can be expanded to provide up to 80-channels of remote control. It is easier to service and has a quick disconnect panel which allows removal of instruments without disturbing wire connections. On-site, one-person calibration is an additional feature. Also on display is TFT's precision FM modulation monitor and turnable RF preselector. This monitor is designed for direct connection to the transmitter and is equipped with a digitally-settable peak flasher which displays plus and minus modulation peaks simultaneously. The preselector allows off-air monitoring of any four FM stations.

Potomac Instruments, Inc. offers for the first time an innovative audio test system which facilitates the measurement of critical parameters in monophonic and stereophonic audio equipment. The test system includes an audio analyzer and an audio generator. The system is designed primarily for commercial broadcast proof-of-performance measurements and equipment maintenance. Both units are RFI shielded to enable accurate measurements in high level radio frequency environments type to broadcast transmitter facilities.

MicMix Audio Products, Inc. makes its entry into the panel meter market with the Master Audio Meter, a dual-channel LED bar-display unit having exceptional features which should prove to be a significant step in obtaining highest performance from consoles and other recording and broadcast equipment. The adjustable brightness display is selectable for either Peak or RMS, is readily switched to either of two independent and externally adjustable reference levels, permitting exact matching to more than one recording device without re-calibration. In addition, a new audio delay unit and special effects generator is among the firm's many products.

Tentel is displaying a new and unique in-line Tension Gauge for use with ½, ¾ and 1-inch helical scan video recorders. Proper tension is critically important for reducing video picture skew and for interchangeability between similar machines. The Tentelometer provides a fast, accurate method of measuring the tension to enable corrective measures to be taken.

Sound Technology has on display a new distortion measurement system which includes a distortion analyzer and oscillator simultaneously tuned in one easy-to-use system. This product, which has been on the market for less than a year, is now becoming accepted by the industry as a standard measurement instrument.

Video Aids Corporation features a new NTSC Test Signal Generator and a multi-line VITS inserter. Since no additional details were made available, I suggest that you stop by its booth for further information.

L-W International features its new Athena model 5000 television projector. This is a broadcast quality projector which incorporates many professional features not available on current industry accepted models. In addition to standard features, the 5000 offers production capabilities for direct video tape recording from film without need for optical printer, animation stands, video disc recorders and special effect devices.

The Sheraton Park . . .

Moving across the street to the Sheraton, we find:

Collins Commercial Telecommunications Division. . . . Collins Radio to us old timers . . . has some new products on display. A 25KW FM Generation 4 Transmitter is compatible with Automatic Transmission Systems (ATS) operation and offers such standard features as automatic power output control, automatic filament voltage regulation, automatic overload recycling, and overload fault indicators. A new 5KW AM Power Rock Transmitter compatable with ATS and AM Stereo is also on display. A new Q-Taper RF output network reduces cross modulation from nearby stations. Collins also offers a new super power circularly polarized FM antenna which incorporates a number of unique features. Also on display is a medium power circularly polarized FM antenna which utilizes a number of proven design and construction techniques employed in the super power series. A new 8-channel stereo console accommodating 26 stereo input pairs also is featured.

Harris Corp. has new products which cannot be adequately treated in such a short presentation as this. However, let me note briefly several of the new products which should be seen at the Harris booth. There's an FM exciter, a 25KW VHF low band color television transmitter, a solid state 1KW AM transmitter applicable to ATS, and a 2-½ KW FM transmitter which incorporates many unique features and is also applicable to ATS. Also on the new list from Harris is an FM audio processor with unequaled versatility, and a PMP circularly polarized self-supporting FM antenna. Again, please stop by the Harris booth for further information on these devices.

International Tapetronics Corp. has four new products. The ITC Thousand Cartridge System is a revolutionary new concept for handling audio tape cartridges and will provide instant access to up to 1024 carts. Another new product is an Erase/Splice Locator for audio tape cartridges which combines both erasing and locating the splice in just one step. Their third new product is an open reel Recorder/Reproducer with a comprehensive list of features. Finally, it has a new digital up-down counter that provides an elapsed, real-time display for tape cartridge systems.

Listec has for your inspection the Vinten PortaPed which is a portable pneumatic pedestal with many practical features. This device must be seen to be appreciated and is a must-see for anyone in the market for a pneumatic pedestal.

Thomson-CSF Laboratories is introducing a Digital Noise Reducer and the Microcam, an 8-pound color TV camera ideal for ENG usage. The Noise Reducer will be demonstrated using marginal video tape and a live

Microcam operating at light levels previously thought impossible with standard ⅔ plumbicons. The Digital Noise Reducer will also be the subject of a paper to be given at the Tuesday morning television session. The Microcam is an 8-pound camera with a 3-½ pound electronic pack.

The Washington Hilton . . .

Last but not least, we'll concentrate now on the Hilton exhibits which are assigned basically to the television management segment of the Convention.

RCA has a number of new products on display, but I'll mention only a few. One is a new high quality compact color TV camera for studio and field use. The TK-760 offers a high degree of stability for field use and the flexible control system needed for any studio or field application. It uses a new ⅔-inch pickup tube, the improved Saticon, recently introduced by RCA Electro-Optics and Devices. Also displayed is a new five-kilowatt completely solid-state AM transmitter using transistor arrays in place of vacuum tubes for reduced power consumption and inherent reliability and minimum maintenance requirement. Also featured at the booth is a complete ENG system, including camera, video tape recording and editing equipment, time base corrector, frame synchronizer, microwave equipment, and two-way radio portable and mobile communications. A new simplified built-in editing device for RCA's TR-600 quad video tape recorder is being demonstrated for the first time. Called the SE-1, the editor provides previewable editing capability and is ideal for commercial "tags", station promos and other simple editing requirements. Finally, RCA is offering three circularly polarized television broadcast antennas which are being displayed in RCA's exhibit, including a Fan-Vee, top mounted unit. Two of the CP antennas are new designs, being shown for the first time. The company also is introducing a new light-weight aluminum pylon antenna for side mounting on a tower.

Ampex Corp. has four new products, including an Electronic Still Store (ESS), the result of a joint engineering venture by Ampex and CBS. It represents a first in broadcasting — use of computer disc technology allowing for high density storage, rapid access and reliability. Also on display is the Ampex ATR-100, a truly professional tape recorder available in four-speeds. It comes in a cabinet but also can be rack-mounted or installed in a portable case. It has a universal power supply and can be used anywhere in the world. The Ampex ATR-700, successor to the 500 and 600 series, also is offered. Full-track or 2-track, stereo, and ¼ track head configurations are available along with two speed pairs. Improved control and speed accuracy and synchronous reproduce features are standard. A new member of the Ampex family of helical VTR's is the new VPR-10. It has all the unsurpassed recording characteristics of the VTR-1 but is a smaller, portable package.

Data Communications Corp. has the new BIAS Automatic Switching System which is now being installed at WNAC-TV, Boston, Mass. This new system is expected to have a substantial impact on the television industry and I would suggest that you visit BIAS for further information.

Fuginon Optical is introducing two new lenses. The 14 × 10CERS f/1.9 for ⅔ inch ENG with a built-in 2X extender which gives it a 544 equivalency. Its new 30 × 20ESM, a utility lens, has a built-in 2X range extender allowing focal length between 20 and 124mm.

Angenieux Corporation of America is displaying a new 15×9.5 total zoom lens system which is of special interest to the electronic journalism segment of the industry. This system consists of the basic 9.5-14mm, f/1.8 zoom lens with a series of accessories which provide extreme flexibility heretofore never achieved with a single lens.

Cetec Broadcast Group is unique in that it has booths located in both the Washington Hilton and the Sheraton Park. Its Sheraton Park booth displays a full line of FM antennas, while a full line of TV antennas is on display at the Hilton. There are also a number of other intriguing devices which are new to the industry and I would suggest that you stop by both booths for further information.

The Olesen Co. is exhibiting a new special effects projector. Having been introduced in the United States but a short time ago the projector will be of special interest to TV designers, directors, and production managers. It offers great flexibility and diversity for lighting and set design.

Telescript, Inc. has on display a new Monitor Prompting System which elminates many of the deficiencies associated with mounting prompting devices on expensive TV studio cameras.

In Conclusion . . .

Needless to say, the foregoing has been but a brief overview of some of the new equipment which can be seen in the three exhibit halls. I hope it has provided you with a starting point for your tour through what we believe is the largest display of broadcast equipment ever assembled anywhere.

As a concluding note, one of the newly emerging innovations which will have a substantial impact upon the AM broadcast service is AM stereo. For those interested, may I suggest that you attend tomorrow morning the AM stereo Workshop which begins promptly at 8:00 a.m. in the DIPLOMAT ROOM of the Shoreham Americana Hotel. It will provide you with an excellent insight of what will be developing in AM broadcasting.

An added reminder — there will be no engineering sessions scheduled for Tuesday afternoon. This is to give all conferees an opportunity to visit the exhibit areas without worrying about missing any of the papers. It's a golden opportunity to zero in on those areas of interest.

Engineering Advisory Committee Report

Robert W. Flanders
Vice President for Engineering
McGraw-Hill Broadcasting Co., Inc.
Indianapolis, Ind.

As Chairman of the NAB Engineering Advisory Committee, I would like to submit the following report on this committee's activities for the year 1976-77. This again was a very active year for the committee with some especially pressing industry problems, plus the normal number of everyday situations.

TV Subtitling . . .

Several comments have been filed in this proceeding (FCC Docket #20693), expressing many proposed uses for the entire blanking area. These include a thorough analysis by the Joint Committee for InterSociety Coordination (JCIC) on TV Broadcast Ancillary Signals.

The Commission has responded to the petition (Docket #20693) of PBS requesting Line #21 be reserved for the use of captioning for the hearing-impaired by making Line #21 available but on a voluntary, not a reserved, basis.

WARC '77 & '79 . . .

The FCC's Third Notice of proposed Rule Making (Docket #20271) relating to preparation of the General World Administrative Radio Conference of the International Telecommunications Union to consider revision of the International Radio Regulations was determined to have in-depth implications as it relates to AM/FM/TV and auxiliary broadcast services.

The committee recommended that the National Association of Broadcasters petition for a 90-day extension of time to file and that a mailing be made to the TV membership advising them of the impact the proposal would have on the UHF-TV bands. The Association was requested to file appropriate comments.

Proposed New Rules for Non-Commercial FM Broadcast Stations

The committee recognizes the deletorious effect of the proposed rule (Docket #20735) on the direct reception of Channel 6 television stations and the potential for increased harmonic interference to other TV channels. It approved comments to be filed urging protection of the reception of TV stations from such interference.

VHF Drop-Ins . . .

The committee noted that this important proceeding (Docket #20418) is in the hands of the FCC and due for action momentarily.

Automatic Transmission Systems (ATS) . . .

The FCC has taken limited action on the ATS proposal, (Docket #20403), which NAB recommended.

The Commission is permitting the automatic transmission systems concept to be used at all FM and non-directional antenna AM stations. It is continuing to study the proposal for future TV and directional AM station use and promises a resolution on this point this year.

AM/FM Receiver Performance Standards . . .

The committee has expressed concern about the quality characteristics of the presently available AM/FM receivers. A special subcommittee has been appointed and charged with the responsibility of developing recommended receiver performance standards and practices.

Balanced Audio Levels . . .

The Subjective Loudness Subcommittee reported on its studies, stating that it finds an important and continuing need for further research to resolve the difficulties associated with this subject.

After substantial discussion the committee decided that due to the importance of this effort and the upcoming retirement of the subcommittee chairman the present Subjective Loudness Subcommittee should be dissolved and the responsibility of further study should rest with the full membership of the EAC.

The committee thanked Chairman Ernie Adams and members of his subcommittee for their dedicated activities and wished Mr. Adams well in his retirement.

Citizens Band . . .

The committee expressed concern about the increasing interference to radio and TV caused by the use of CB and encouraged the staff to participate in all activities intended to resolve this matter. (Docket #20120).

Quadraphonic FM . . .

Adoption of standards for FM stereo-quadraphonic transmission is provided in RM2742. After careful study, the committee decided not to participate in this effort.

AM Stereo . . .

The committee notes that the National AM Stereophonic Radio Committee is making progress and that five systems are under active consideration with field testing to begin this spring.

Recording Standards . . .

The new NAB Cart Recording and Reproducing Standard has been submitted to the International Electromagnetic Committee (IEC) for adoption as an international standard.

The NAB reel-to-reel Tape Standard Subcommittee is being reactivated with review and updating of the reel-to-reel standard as its goal.

Headquarters Technical Laboratory . . .

The committee reviewed a staff proposal to install and equip a technical laboratory at NAB headquarters to assist in monitoring and/or testing technical parameters of the various dockets before the FCC. It was decided that this proposal should be given further study and is to be reconsidered at the next EAC meeting.

Seminars . . .

The Directional Antenna Seminar has been well attended and the Engineering/Management Development Seminar has now graduated more than 500 conferees.

This concludes the EAC report.

I wish to thank the following committee members who have devoted their time to this project.

Charles F. Abel, Manager of Engineering, KFMB San Diego, Cal.

Ernest L. Adams, Vice President for Engineering, Cox Broadcasting Corp., Atlanta, Ga.

Ralph F. Batt, Vice President & Manager of Engineering, WGN, Chicago.

Albin R. Hillstrom, Vice President for Engineering, KOOL Radio and Television, Phoenix, Ariz.

Martin Meaney, Director, Allocations Engineering, NBC, New York.

James D. Parker, Staff Consultant—Telecommunications, CBS Television Network, New York.

R. LaVerne Pointer, Director, Broadcast Engineering, ABC, New York.

William Wisniewski, Director of Engineering, MBS, Washington.

Benjamin Wolfe, Vice President for Engineering, Post-Newsweek Stations, Washington.

And a special thanks to A. James Ebel, President and General Manager of Station KOLN-TV, Lincoln, Nebraska, NAB Board Liaison Member.

Progress Report — JCIC Activities

Roland J. Zavada*
Vice President, Engineering
SMPTE

The Joint Committee on Intersociety Coordination, representing the EIA, IEEE, NAB, NCTA, and SMPTE, meets upon call from any of the member organizations to discuss specific subjects which may involve the other members and may warrant joint action, either by a statement of policy or initiation of an appropriate committee activity.

Of the three Ad Hoc Committees set up since 1968, the Committee for Color Television Study has completed its work, the Committee for Television Sound has been relatively inactive, but continues its work, and considerable work has been done by the Committee on Television Broadcast Ancillary Signals.

A meeting was held in October to review the progress of the JCIC and to consider several new items. Significant among these were:

- A recommendation by NCTA to provide coded program identification for automated non-duplication switching. This subject will require further evaluation and study before separate committee activity is initiated or work assigned.
- The need to expand the Ancillary Signal Committee's scope to incorporate the study of high data rate systems such as Teletext was reviewed. NAB was assigned the task of advising the committee whether broadcasters could anticipate a need for such a system in their commercial broadcast applications. Any future testing or study, therefore, depends upon NAB's assessment of broadcaster needs.
- A recommendation was made by SMPTE that a study be undertaken to evaluate the potential and interest in high-resolution television systems. SMPTE was given the responsibility by the parent committee and has set up a study group with Mr. Donald Fink as Chairman.

By way of review, the JCIC Committee Broadcast Ancillary Signals on Television was established to identify the time and frequency domains within the television program signals that are technically feasible for the inclusion of special signals; to study the unique features and limitations applicable to these domains; to establish the priorities to be assigned the various functions that could be accomplished by these special "piggybacked" signals, and to recommend to the FCC a set of guidelines against which ancillary signal proposals could be evaluated.

This Progress Report covers the Committee's activities during the past year.

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*Prepared in cooperation with Committee Chairmen **Robert O'Connor, Richard Barton** and **Daniel Wells.**

Sub-Committee on Audio Systems

Our work in the area of audio ancillary signal systems resulted from a request by the FCC that an industry committee be set up to look into coding systems that would be compatible with both motion picture film and video tape.

SMPTE was asked to assume responsibility for this work as a subcommittee and responded by establishing the SMPTE Working Group on Aural Program Identification Systems. It currently is chaired by Mr. Richard Bartow.

An extensive patent and literature search was undertaken with the result that some 15 methods of conveying coded information in audio signals were identified — with six of them potentially applicable to the area of concern. Because FCC, in its report and order, specifically requested that the aural system proposed by Audicom Corporation (under Docket #18877) be given full consideration, activities of the working group to date have used this particular system as the focus of its evaluation.

The Audicom system incorporates the use of subaudible frequency shift keyed (FSK) tones to insert the necessary code sequences into the program audio. The signal level of the FSK tone varies from 40 to 55 dB below nominal program level. Actual insertion of the code sequences is accomplished by using a narrow-band "notch" during the code interval to delete program audio content around a center frequency of 2877 Hz.

With the assistance of staff members of the Eastman School of Music in Rochester, N.Y., a series of musical works and voice recordings was selected. A composite master recording was supplied to Audicom for use in making an encoded master.

Following a successful demonstration of the Audicom system using this encoded master, arrangements were made in cooperation with the ABC Broadcasting Network to obtain return signal recordings by transmitting the encoded signal around its radio and TV round-robin networks. Other test recordings were obtained by making copies of the master encoded tape on high-speed duplicators and through optical sound transfers on motion picture film.

This test was designed to determine the ability of the Audicom system to properly detect coded information contained in "processed" sound recordings representing commercial sound practice.

When subjected to decoding, the results of this series of test sequences were not too encouraging. The tests indicated that although the system meets the prime requirement of ancillary signals, namely that no significant degradation be caused to the program signal, the observed *reliability* of the technique proved to be inadequate because of deficiencies in the prototype hardware. The hardware previously had demonstrated successfully with the reference tape.

These initial tests have pointed up the real problem involved in meeting the two conflicting requirements of any ancillary signal system involving multiplexing within the program signal:

1. The added signal must be of low enough amplitude, for an audio system, or small enough in terms of the occupied portion of the raster in terms of video systems, so as not to be heard or seen.

2. It must be of sufficient magnitude and possess other "survival" features to provide a *reliable* service — under all conventional modes of operations, including transmission over thousands of miles of network interconnection facilities.

As necessary background for addressing the general application of FSK techniques to the area of inserted audio codes, extensive work was done to evaluate the effects of instantaneous velocity variations in the transport mechanisms of telecine motion picture projectors and 2-inch quadraplex video tape recorders. A formal report covering this work was prepared by the Working Group and is contained in the 1976 NAB Proceedings.

The information in this report is presented in a generalized format to enable system evaluation based on the probability of error occurrence as a function of system frequency shift and the frequency response characteristics of associated system circuitry.

Work is continuing to assess the velocity variation characteristics and major flutter components of magnetic tape transport mechanisms. Twenty-seven manufacturers/suppliers of transports have been identified and contacted for information on one and two-inch reel-to-reel, two-inch cartridge, and ¾-inch video cassettes and ¼-inch reel-to-reel and cartridge audio tape transports.

To date, very little information has been available on magnetic tape transports. The working group learned of similar velocity variation studies are being undertaken by the BBC, and is currently corresponding with them.

Listening tests to quantify hearing response to the presence of inserted codes in various kinds of program audio are in the final planning stage. Because it now appears that the Audicom or similar technology may have application in the area of copyright protection of sound recordings used in broadcasting, the results of the committee's testing to assess the degree of program degradation should be useful to the commission in its further considerations of the Audicom docket.

Vertical Interval Systems . . .

Several members of the Committee, whose affiliated companies had prepared petitions for rule-making involving ancillary signals, presented these petitions to the committee for its comments. The proposals involved three quite different systems:

1. *PBS proposal for the use of Line 21 for a program captioning signal for the hearing-impaired.*

This proposal involved the first suggestion for a system involving the broadcast of alpha numeric information to home receivers, and was the subject of considerable study by the committee. The committee's consensus was presented to the FCC in the form of a Statement Relating to RM-2616 and subsequent Comments in response to the Notice of Proposed Rule-Making (Docket 20693).

The committee noted that systems involving alpha numeric (and graphics) transmissions in the vertical blanking interval are currently under study in at least six other countries throughout the world. But it noted that these systems involve a higher transmission rate of information, and are intended for the entire viewer population, including the hearing-impaired community.

The committee has studied these systems — which have come to be known by the generic term "Teletext." From a *technical* point of view, the committee recognizes that such systems hold the potential for providing a wide variety of program-related and broadcast-related services in a highly efficient manner.

The committee pointed out in its comments that the application of such a technique to the U.S. system of television broadcasting, would require an extensive study of such items as: optimum transmission bit rate, appropriate page replenishment rates, acceptable cost versus spectrum compromise, and others. All of these parameters would have to be studied in terms of both network-originated, and locally-originated services.

The committee also recognized that many other *non-technical* factors are involved, such as the need for such a system, and its economic viability. The committee in its comments indicated its intention to seek, and has since sought, the guidance of the parent JCIC committee as to whether or not such a technical study should be initiated. The parent committee has charged NAB, the sponsoring association for the Ad Hoc Committee, with the resolution of this question.

2. *NBC proposed petition for digital network source identification signal on line 20, field one.*

This proposal involves a 48-bit digital signal containing the following information in binary form: network, location of program origination, and the month, day, hours, minutes and seconds of program origination time.

This signal is regarded as a "professional use" data signal for use within the industry and not by home viewers. In addition to enabling a more accurate and timely determination to be made of network program carriage by affiliated stations, the signal has potential applicability to three other functions that were listed in the charge given to the committee, that of network signalling, equipment cure signal, and automatic operation of CATV non-duplication switchers.

With possible subsequent expansion, the additional function of automatic program logging might be included.

3. *ABC proposed petition for one-line-per-field VIT signal.*

Current FCC requirements dictate that remotely-controlled television broadcast stations transmit prescribed vertical interval test signals (VITS) in three prescribed positions within the vertical blanking interval with the use of a fourth position optional.

The ABC petition would revise this requirement to specify only *two* positions — specifically, line 17 of both fields, with the required signal made identical to the signal currently used for network transmission quality surveillance by all of the networks.

The committee has since its inception stated its belief that the Commission's objectives could be met with only a single-line-per-field VIT signal. It further believes that, in view of the advent of automatic VITS monitoring and correction equipment, a unified signal for the monitoring and/or correction of both network transmission facilities and broadcast transmitters is highly desirable.

The committee expressed its support of these two proposed concepts and suggested that the minor differences of opinion that exist with respect to the format of the current signals be resolved by the three specific permanent committees involved in such matters: the Network Transmission Committee, IEEE 2.1.4 and EIA TR-4.1.

Delegated Tasks . . .

In accordance with one of the charges given to it, the Ad Hoc Committee has delegated several specific projects to existing permanent industry committees for resolution, such as the format of VIT signals as described above.

Another of these projects is the consideration of a system equalization signal and an associated adaptive equalizer that could correct for multipath distortion.

The program-related vertical interval reference (VIR) signal provides for the correction of color variation among different program segments and among different stations. The locally-generated vertical interval test (VIT) signal provides for the monitoring of transmitter performance — and provides the potential for the automatic *correction* of transmitter deficiencies.

The one missing link at the present time is a method for the correction of multipath distortion or ghosting caused by the propagation patch, over which the broadcaster has no control. (This propagation path is defined as the "air link," the receiving antenna, the transmission line to the receiver, and the receiver termination).

This effort has been delegated to EIA's Broadcast Television Systems Committee. In this study, consideration will be given to a new vertical interval signal, as well as to the possibility of using a portion of the VIT signal as a basis for the adaptive time-domain correction.

Current Plans . . .

During the coming year, the committee hopes to:

- Complete its studies of the use of the program audio signal for ancillary signals.
- Conduct a test to determine if vertical lines other than 17 through 21 are technically feasible for ancillary signals.
- Pursue whatever course of action that may be directed by the parent JCIC Committee in connection with the proposed "Teletext" study.

The committee intends to follow its first and second Interim reports with a Final Report that will:

(1) Summarize its activities.

(2) Review the pros and cons and other considerations involved in the use for ancillary signals in the horizontal blanking interval, the vertical blanking interval, the program audio, and the program video signal.

(3) Recommend a master line allocation plan for the vertical interval.

(4) Recommend a set of guidelines against which future ancillary signal proposals may be evaluated.

Study of Television Sound . . .

In review, the purpose of the JCIC Ad Hoc Committee for the Study of Television Sound is to examine the entire television sound system from the studio origination to the sound heard in the home; to identify problems and opportunities for improvement, and to assign tasks to appropriate organizations to accomplish these improvements.

The organization of the committee and a detailed description of their scope and objectives was reported in last year's NAB proceedings, by Daniel R. Wells, the Ad Hoc Committee Chairman. A detailed status report appeared in the June 1976 issue of the **SMPTE Journal.**

All aspects have not been covered to date, and some assignments remain. However, work is progressing in the six panels of production, distribution, broadcasting, reception, cable, and master antennas and state of the art techniques.

Progress is anticipated during the coming year on:

A. The correlation between the amount of incidental phase modulation at the transmitter with audio SNR for the home viewer.

B. Audio subcarrier modulation deviation and whether double or triple the present ±25 kHz deviation can be achieved for current circuitry.

C. Possible further testing of 75 s versus 25 s preemphasis curve and the relationship of limiting at the television transmitter.

D. Whether there should be a "standard" for peak reading meters?

E. How to test harmonic distortion in audio channels composed of multiplexed telephony circuits.

Some of these projects are in the definition stage and Mr. Wells would appreciate any comments or suggestions you may have.

Color Television Study Committee . . .

The Color Television Study Committee was organized in 1968 under the administration of the SMPTE. The charge to the group by the JCIC was to single out causes for variations in color television pictures viewed in the home, particularly as regards to variability in hue, saturation, and color quality.

During its term of eight years, the work touched on all elements of production, reproduction, broadcast transmission to reception with the home receiver.

Basically, the role assigned by JCIC has been fulfilled and the committee chairman, Mr. K. Blair Bensen, recommended it be disbanded with all work assigned.

Responsibility for color transmission standards has been assigned to the EIA BTS Committee, and work is continuing on the video wave form specifications (TR4.4.2).

Monitor standards was similarly assigned and work is continuing under EIA TR 4.4.2.

Most of the work has been directed toward the problem of colorimetric uniformity in film and live origination and with color fidelity of display devices.

Several years ago this activity was assigned to the SMPTE and is organized as a subcommittee under the Television Video Technology Committee with Mr. Leroy DeMarsh as chairman.

The FCC rules and regulations are based upon NTSC phosphors. Because phosphors, in current commercial use, differ significantly from NTSC specifications and because the EBU phosphor recommendations also differ, a very extensive study was undertaken for the JCIC committee to determine if the NTSC standard required modification.

The consensus derived and reaffirmed is that we retain the NTSC phosphor standard and take steps or undertake studies to bring current commercially available monitors into conformity.

Conclusions . . .

The 15 minutes or so taken to make this report inadequately documents the multiple hours of dedicated effort of the committee participants and the leadership of the ad hoc and subcommittee chairmen working to our mutual benefit.

In acknowledging our appreciation to them, we also recognize the support of their sponsoring companies in providing time, travel and understanding cooperation.

Our work continues. . . .

A New Standard (in preparation) Governing the Performance of Television Broadcast Demodulators

T. M. Gluyas
Chairman,
TV Demodulator Standards
Task Force
Staff Engineer,
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A need for television demodulator standards has been felt, generally, for quite a few years.

Broadcasters have observed that test waveforms and picture quality monitored through a demodulator often are not, as expected, based on other transmitter test measurements. Sometimes, a good picture at the demodulator output does not seem to correlate very well with transmitter adjustments that yield the best picture at home.

The public, but more especially TV receiver design engineers, have decried the lack of uniform good picture quality among stations. Part of this is believed to be due to station adjustments made with different demodulators in the test loops.

Consequently, the EIA Broadcast Television Broadcast Systems Committee (BTS) decided to do something about this and in January 1975 organized a Task Force to draft a proposal for Electrical Performance Standards for Television Demodulators.

Evolution of the Standards Proposal

The Task Force was set up to include representatives from domestically manufactured or distributed TV broadcast demodulators, TV receiver manufacturers, TV transmitter manufacturers, a network broadcaster, an independent broadcaster, and a TV cable company.

In February, 1975, the BTS Committee drafted a "Charge to the Task Force" which was both a work statement and a preliminary outline for a Standards Proposal. The outline includes several operating modes for the demodulator.

The work statement has been followed by the Task Force with little deviation except for standards on a demodulator mode intended to be "representative of demodulation characteristics of current and near future TV receivers."

Work on standards for that mode was carried quite far before it was decided that its limited usefulness would not justify the considerable added cost and complexity of the additional mode. At that point it was dropped.

The circuit requirements are quite different for a TV receiver that normally has separate luminance and chrominance signal channels and a demodulator that must have an overall flat response to deliver combined luminance and chrominance signals to existing waveform and picture monitors.

The Task Force began work at a meeting on March 25, 1975—just two years ago—and has held 11 meetings at approximately bimonthly intervals since that date.

The first task was to accumulate a large amount of carefully measured performance data (and predictions for next generation equipment performance) on TV receivers, transmitters and demodulators.

A draft of the standards proposal is complete at the present time but it is subject to change during review by the Task Force, the BTS Committee, and other EIA hierarchy.

Demodulator Functions . . .

Television demodulators are employed in measuring transmitter performance and to facilitate transmitter adjustment. They also are employed in assessing radiated signal waveforms and picture quality.

Some tests may be carried out under conditions where the visual transmitter can be operated without the aural (wideband demodulator mode) but many cannot. These require a sound notch or bandwidth limited mode.

It also is often necessary or desirable to monitor performance from a remote site and this requires a high sensitivity mode with a sound notch. However, some tests cannot be made with precision from a remote site.

This leads to performance standards for several operating modes and two basic input signal levels—a high sensitivity input for remote use and a low sensitivity input for transmitter site use. Consequently, performance standards have been drafted for each of the following modes:

1 Wideband, synchronous detection
1A Synchronous detection, quadrature output
2 Wideband, envelope detection
3 Bandwidth limited (sound notch), synchronous
4 Bandwidth limited, envelope detection

In setting performance limits for the many specifications covered in the proposed standard, it has been the Task Force philosophy not to set limits much tighter than a level which might produce just noticeable picture impairments.

Performance limits on the other hand, have been chosen to be tight enough to permit precision measurement of transmitter performance—in other words, tighter than equivalent transmitter specifications. This was not possible in every case because of equipment cost-complexity considerations, or lack of available means for proving compliance of the demodulator to the stringent performance standards desired.

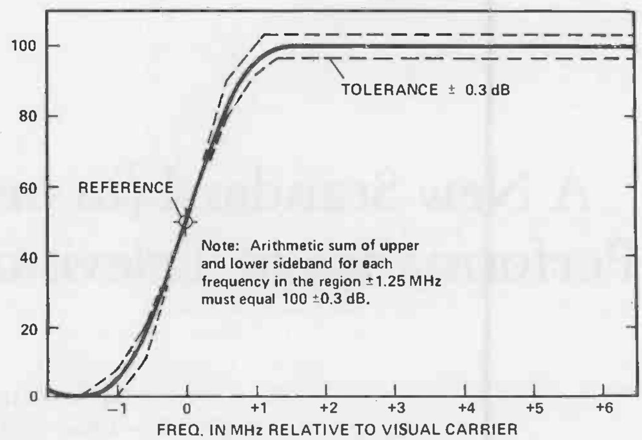


Figure 1.

Amplitude vs. Sideband Frequency Response

Figure 1 shows the amplitude vs. sideband frequency response of the demodulator in the wideband mode.

The Nyquist slope received a great deal of attention. It departs slightly from the classic "ideal vestigial demodulator."

The slope was made as gradual as possible in order to receive, and be influenced by, the maximum useful lower sideband of the TV transmitter, but no so gradual as to receive frequencies lower than this, or out-of-channel frequencies.

Excessive lower sideband response could produce unrealistically good results on a double-sideband test generator but misleadingly poor results on a good vestigial sideband transmitter.[1]

It is known that some TV home receivers have a more gradual slope than Figure 1. Good envelope delay in an economical manner was necessary, however, and this has been accomplished with acceptably small errors in luminance transient response.[2] However, this compromise was not emulated in the broadcast demodulator standards.

1. T. M. Gluyas, "TV transmitter luminance transient response," IEEE Transactions on Broadcasting, Vol. BC-20, No. 1, March 1974.
2. C. B. Patel and J. P. Bingham, "The effects of vestigial sideband passband characteristics upon television transient response," IEEE Transactions on Broadcasting, Vol. BC-22, No. 3, Sept. 1976.

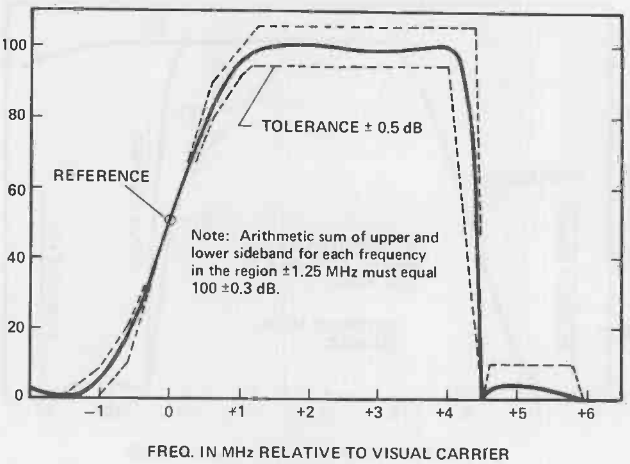


Figure 2.

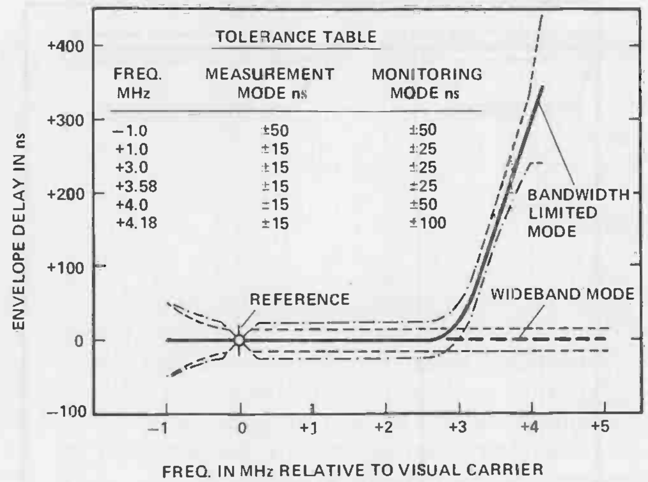


Figure 3.

Figure 2 shows the amplitude vs. sideband frequency response in the bandwidth limited mode. To avoid chrominance transients on colored edges in the picture to the full measure possible within the alloted channel width, the response should be flat, or at least symmetrical, about the color subcarrier frequency over the range 3.58 ± 0.6 MHz. The selected tolerance permits an acceptable departure from the ideal symmetry near the high frequency edge of the chrominance channel.

Envelope Delay vs. Sideband Frequency Response . . .

Figure 3 shows the demodulator envelope delay specifications in wideband and in bandwidth limited modes. In this figure, these modes are called *measurement* and *monitoring* modes which are short titles for the same thing.

A requirement to achieve a bandpass response with comparatively sharp corners as in Figure 2, coupled with tight control of envelope delay as in Figure 1, can lead to filters and delay equalizers with many sections. A consequence of this is rapid fluctuations in phase vs. frequency (ripples in the envelope delay curve) even for small departure from linear phase.

The same problem occurs in the use of surface acoustic wave filters (SAW filters). The measured envelope delay of a typical SAW filter is shown in **Figure 4.**

Since envelope delay is the derivative of the phase vs. frequency curve, the magnitude of the envelope delay errors are directly proportional to the rate at which phase departures occur. This places an unreasonable and meaningless burden on the use of SAW filters and the design of L C filters and equalizers having many sections.

An attempt was made to substitute a more meaningful constraint on demodulator performance by specifying departure from linear phase vs. frequency (phase error) in place of envelope delay.[3] However, no off-the-shelf equipment exists for measuring this quantity with adequate precision.

A less than elegant way out of this dilemma, adopted in the standard, was to employ the specification given in Figure 3, but to expand the tolerance in proportion to the ripple rate above 1 cycle/MHz or to the rate of departure from the specified delay curve.

3. Harold A. Wheeler, "The interpretation of amplitude and phase distortion in terms of paired echoes," Proc. of the IRE, Vol. 27, No. 6, pp 359-385, June 1939.

Output Amplitude vs. Modulating Frequency . . .

This specification involves a combination of a double-sideband test signal generator and the demodulator. It is a "video-in to video-out" or "video-to-video" test.

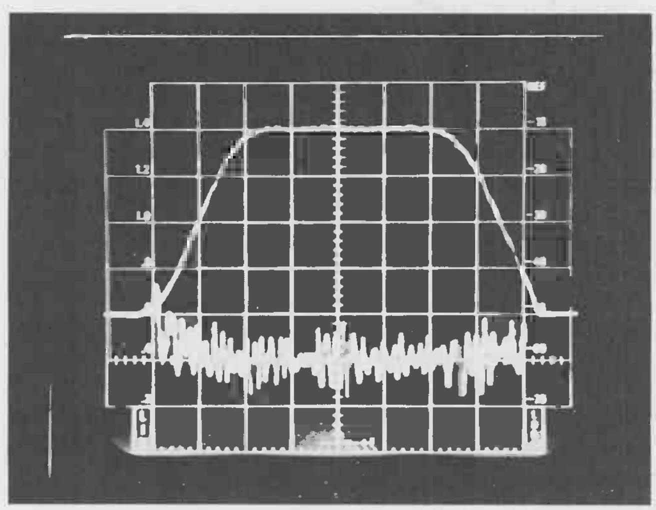


Figure 4.

SAW filter mm envelope delay (bottom trace). Horizontal = 1 MHZ/div., vertical = 50 ns/div. Similar high rates of envelope delay ripple occur in multisection, lumped-constant filters and equalizers.

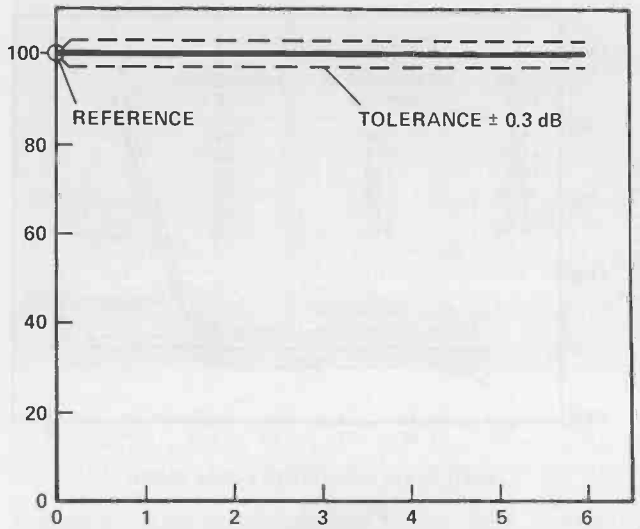


Figure 5.

The performance specification is shown in **Figure 5**. In a sense, this is a redundant test to assure the symmetry of amplitude response about the visual carrier in the region visual carrier ± 1.25 MHz, as specified in Figure 1. Also, it provides an indirect check on any serious departure of phase linearity in this region.

The "video-to-video" test by itself, with a double sideband signal generator, is not sufficient to test the demodulator performance in the critical Nyquist slope region.

RF + IF Amplitude Vs. Sideband Frequency Response . . .

Since a standard has been included for the overall sideband frequency response of the demodulator, it might seem unnecessary to have an additional standard for the RF + IF response and, indeed, this is true for the synchronous detector modes.

However, envelope detection generates quadrature distortion which is a function of the frequency response of the circuits preceding the detector, mainly the IF circuit.[4] Consequently, it is necessary to have standardized RF + IF response in the envelope detection modes if demodulators are to produce similar test and monitoring results.

Proposed standards for this parameter are shown in Figure 6.

Tolerances, included in the standard, are omitted in this figure simply to avoid clutter. Notice that there is one standard for the wideband mode but two alternate standards for the soundnotch mode. Neither of these curves is similar to the response of a typical TV receiver.

A standardized RF + IF response was one of the major compromises reached by the Task Force and merits some explanation.

Part of the compromise arises from the diverse characteristics of existing demodulators which, no doubt, will

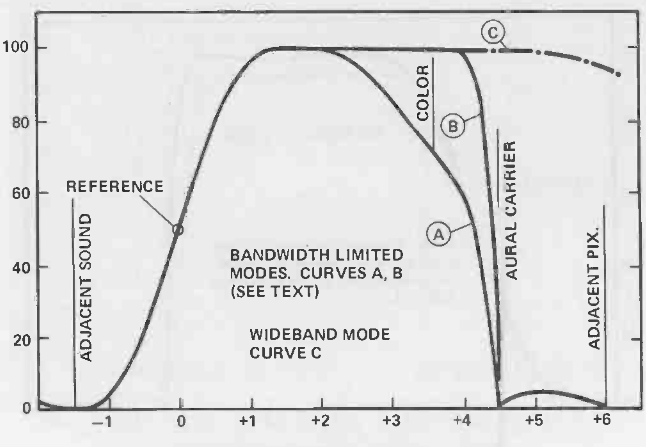


Figure 6.

form the basis of evolution toward improved demodulators.

In the sound notch mode, the IF response of the Harris and Tektronix demodulators are essentially flat as in curve B, whereas the Rohde & Schwarz and Telemet demodulators are rolled off, somewhat similar to curve A, to about − 3.5 or − 4 dB at color subcarrier.

The current Scientific Atlanta demodulator more closely follows receiver practice with an IF response of about − 6 dB at color subcarrier.

There are no U.S. standards and very few international standards that can be examined for precedence. A German standard[5] requires the response to be − 3.5 dB (sound notch in) and a tentative Australian standard[6] specifies − 3 dB at color subcarrier frequency.

In the wideband mode, the Harris, Tektronix and Rohde & Schwarz demodulators have flat IF response, while Telemet retains a response rolled off at color subcarrier.

There are good reasons for these diverse choices and no one choice is clearly best. One response characteristic is superior for one type of signal or test waveform while another characteristic appears to be best for some other type of signal.

Since there is no clearly best overall choice, it was agreed to permit alternate standards to the extent shown in Figure 6, provided that manufacturers will be asked to publish in their instruction books, typical waveforms, obtained with a test signal generator, for commonly employed test waveforms.

The intent is that these waveforms will be used by the broadcaster for comparison to waveforms obtained in TV transmission system tests.

To carry the discussion further, it is necessary to consider how quadrature distortion affects typical test waveforms and picture quality.

4. T. Murakami and R. W. Sonnenfeldt, "Transient response of detectors in symmetric and asymmetric sideband systems," RCA Review, Vol. XVII, No. 4, pp 581-611, December 1955.

5. Technische Pflichtenhefte No. 5/2.2, August 1971. Institut für Rundfunktechnik GmbH (ARD Specification).

6. Characteristics of the Standard Demodulator, Appendix C, Excerpt from Australian Broadcasting Control Board Draft Standards for the Technical Equipment and Operation of Television Station, 1974.

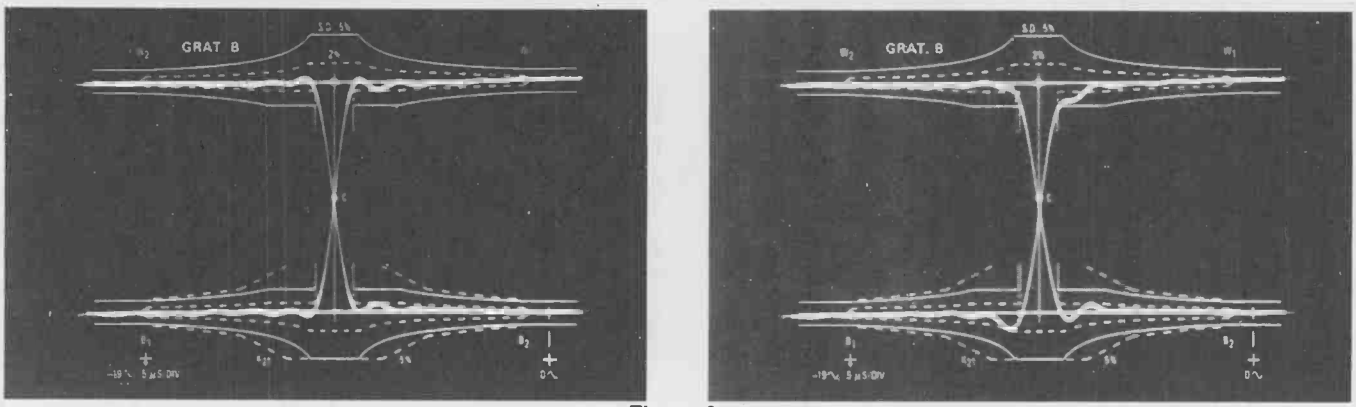


Figure 8.

Comparison between synchronous detection (on the left) and envelope detection (on the right) for normal and inverted 2T sine-squared pulse. Note differences between black level and white level transitions, and effect of pulse polarity on pulse width, for envelope detection.

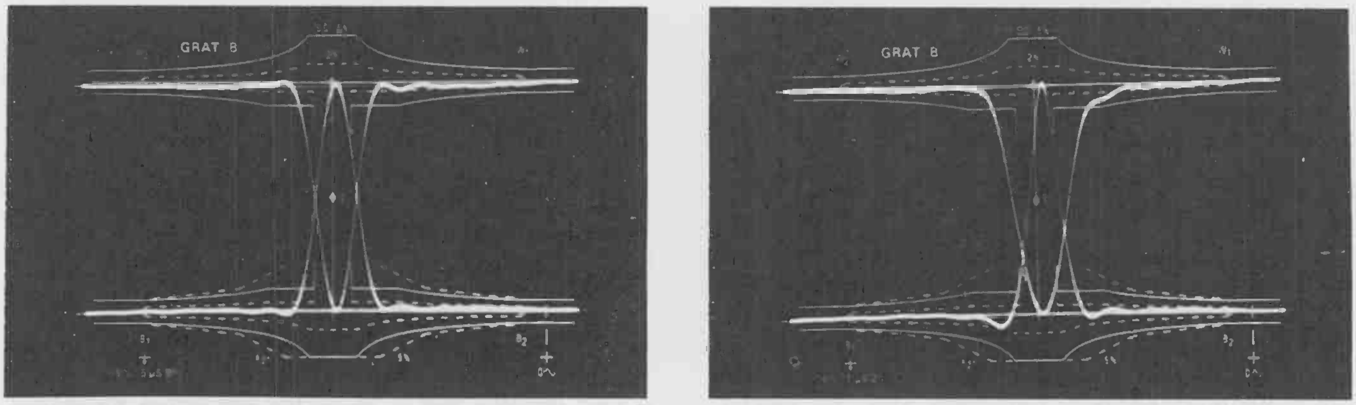


Figure 7.

Comparison between synchronous detection (on the left) and envelope detection (on the right) for leading and trailing edges of 1T sine-squared bar. Note differences between black level and white level transitions and additional waveform distortion for envelope detection.

Quadrature Distortion . . .

Quadrature distortion effects on a full amplitude 2T sine-squared pulse and 1T bar are apparent in **Figures 7** and **8** which compare signals from a synchronous detector and an envelope detector.

If a transmitter low frequency envelope delay adjustment should be carried out observing only one edge of the bar (both edges are superimposed in Figure 8), then the transmitter could be maladjusted to partially compensate for what appeared to be envelope delay errors but were, in fact, errors due to demodulator quadrature distortion.

It is well known that envelope detection quadrature distortion reduces the chrominance and brightness of colored areas in the picture.[7]

Those reductions do not track exactly so that color saturation also is distorted. That is one of several reasons why it is standard practice in current TV receivers to reduce the IF response at color subcarrier by approximately 6 dB.

The following table compares the reduction in the 3.58 MHz chrominance signal and reduction in the luminance

COLOR BAR DISTORTIONS DUE TO QUADRATURE DISTORTION

	Flat IF		IF, – 6dB at 3.58 MHz	
Color	Chrominance* dB	Brightness* dB	Chrominance* dB	Brightness* dB
Red	– 0.3	– 2.8	0	– 0.6
Green	– 0.4	– 1.5	– 0.1	– 0.3
Blue	– 0.1	– 3.4	0	– 0.7
Yellow	– 0.5	– 0.8	– 0.1	– 0.2
Cyan	– 0.6	– 1.7	– 0.2	– 0.4
Magenta	– 0.3	– 1.8	0	– 0.4

*These terms are used here to represent change in 3.58 modulation level and axis shift at transmitter output, — not the actual change in optical output from the color kinescope.

7. W. L. Behrend, "Performance comparison of TV transmitter RF demodulators and the home receivers, IEEE Transactions on Broadcasting, Vol. BC-17, No. 1, March 1971.

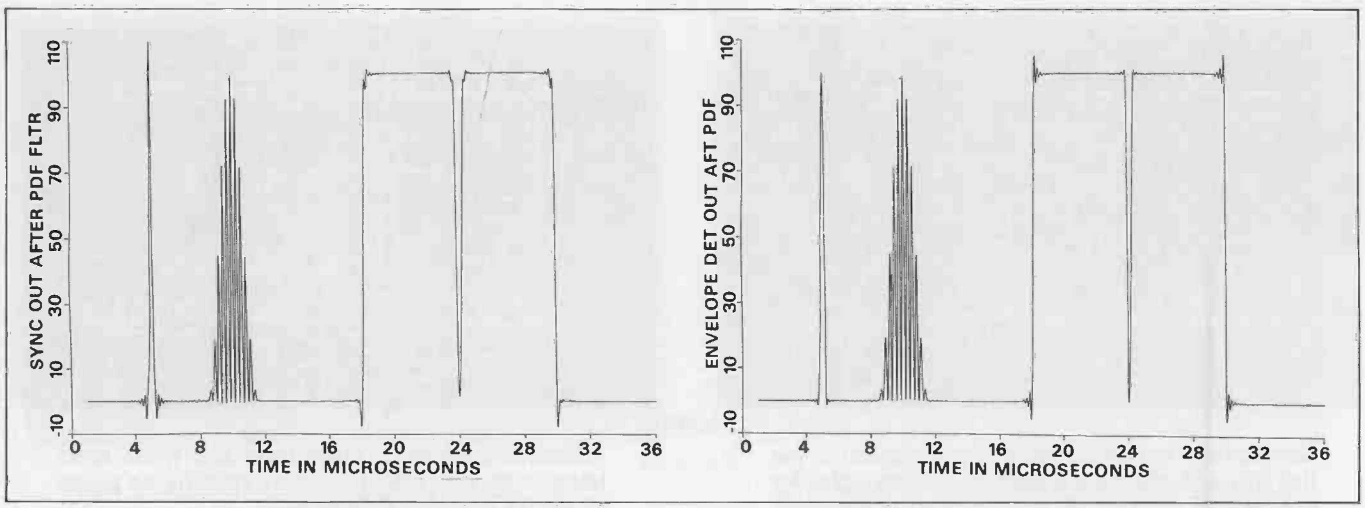


Figure 9.
Analytical comparison between synchronous detection (on the left) and envelope detection (on the right) for a full amplitude 2T sine-squared pulse and 1T bar when the demodulator has a "haystack" IF (response −4 dB @ 3.58 MHz, −7 dB @ 4.18 MHz) and peaked video.

signal for 75% amplitude 100% saturated color bars for a flat IF response and for typical TV receiver IF response.

A direct comparison between a transmitter input picture and an output picture, using a demodulator with flat IF, will show small but observable differences in color brightness and saturation.

It is an interesting fact that color brightness changes can sometimes appear as hue shifts, also—for psycho-physical reasons. Similar distortion probably could not be detected if the demodulator IF response simulated TV receiver response but in that case other forms of distortion would be changed or created for the following reasons.

It is a basic requirement that the overall response of a broadcast demodulator must have substantially flat amplitude vs. frequency response to at least 4 MHz. If the IF response is rolled off, the video response must be peaked to compensate.

Video peaking increases the level of harmonics generated by quadrature distortion for all picture signal components below 2 or 3 MHz and for higher frequencies, as well, unless the video circuits are bandwidth limited. One result of this is a slight increase in high frequency ringing on sharp edges in the picture and a distortion of the pulse-to-bar ratio of the VIT test signal.

Figure 9 exaggerates these effects for illustrative purposes.

The results are for a "haystack" shape IF with no delay error, an amplitude response of −4 dB at 3.58 MHz and −7 dB at 4.18 MHz. To make the overall system flat to 4.18 MHz, the simulated video response was allowed to peak to +11 dB beyond 4.18 MHz.

The curve is one of 146 curves provided by C. Bailey Neal of GTE Sylvania and studied by the Task Force. The curves were computer calculated and computer drawn with great precision.[8]

The Task Force decided that a −3 dB rolloff of the IF at 3.58 MHz in the demodulator monitoring mode with envelope detection resulted in an acceptable compromise between various picture and waveform impairments.

In the synchronous detection mode the RF + IF response above visual carrier is of no particular consequence and no standard was prepared for that mode.

Video Amplifier Response . . .

The Task Force struggled with a standard for video amplifier bandwidth and response.

One decision was to make the test signal waveforms appear as nearly correct as possible on the waveform monitor, in spite of demodulator quadrature distortion, when the transmission system is performing correctly.

Take the simple case of reproducing a multi-burst signal. The multiburst test signal is the simplest case to understand because only single tones are presented, one at a time. If the IF is flat and if the video amplifier is flat, has constant time delay, and is sufficiently wideband to pass all important harmonics, then the multiburst will have correct peak to peak values.

This point is illustrated in **Figure 10** which compares synchronous detection (correct waveform) and envelope detection waveforms.

Superficially, the multi-burst signal will appear to be correct. The waveform distortion will not be apparent unless the monitor time base is changed to view individual cycles as in **Figure 11.**[9]

The axis shift (luminance error) will not be apparent unless the IRE rolloff is employed as in **Figure 12.**

8. S. K. Goyal, C. B. Neal and E. Bowerman, "Television transient response computations using a new simulation program," presented at the 26th Annual Symposium, IEEE Broadcasting Group, Washington, D.C., Sept. 23, 24, 1976.
9. A. G. Uyttendaele and A. H. Bott, "Envelope and synchronous demodulation of V.S.B. TV signals," Proc. of 29th Annual Broadcast Engineering Conference, NAB, Las Vegas, Nevada, April 6-9, 1975.

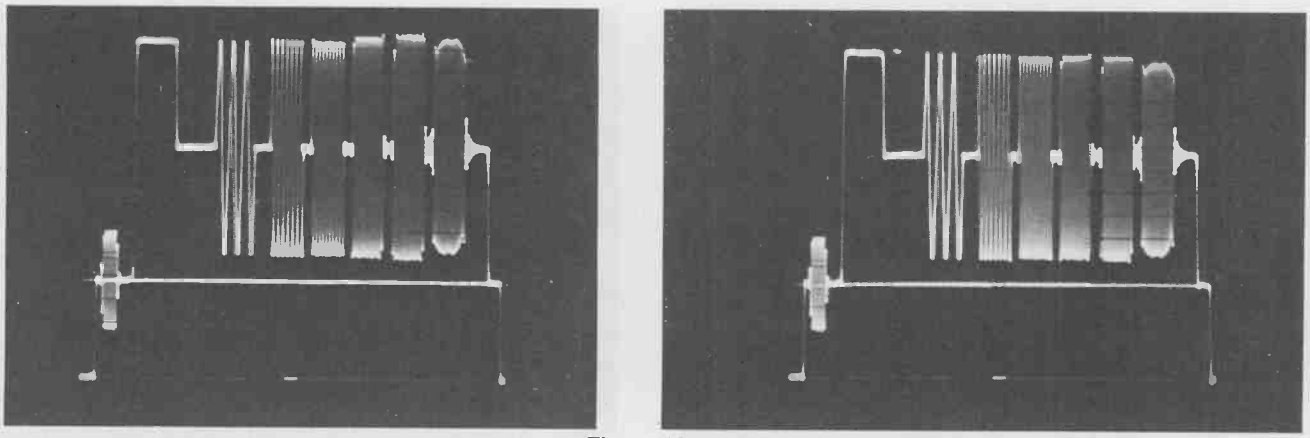


Figure 10.

Comparison between synchronous detection (on the left) and envelope detection (on the right) for a full amplitude multi-burst signal when the IF is flat and bandwidth limited to approximately 5 MHz; and the video amplifier is quite wide-band and phase linear.

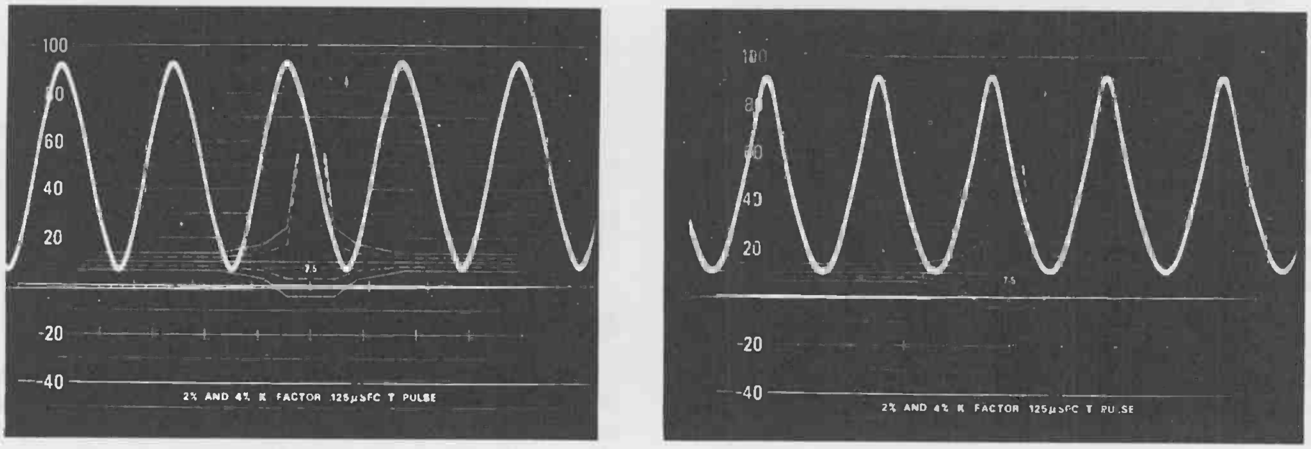


Figure 11.

Comparison between synchronous detection (on the left) and envelope detection (on the right) for the 1 MHz component of a multiburst signal. Note distortion in waveform on the right.

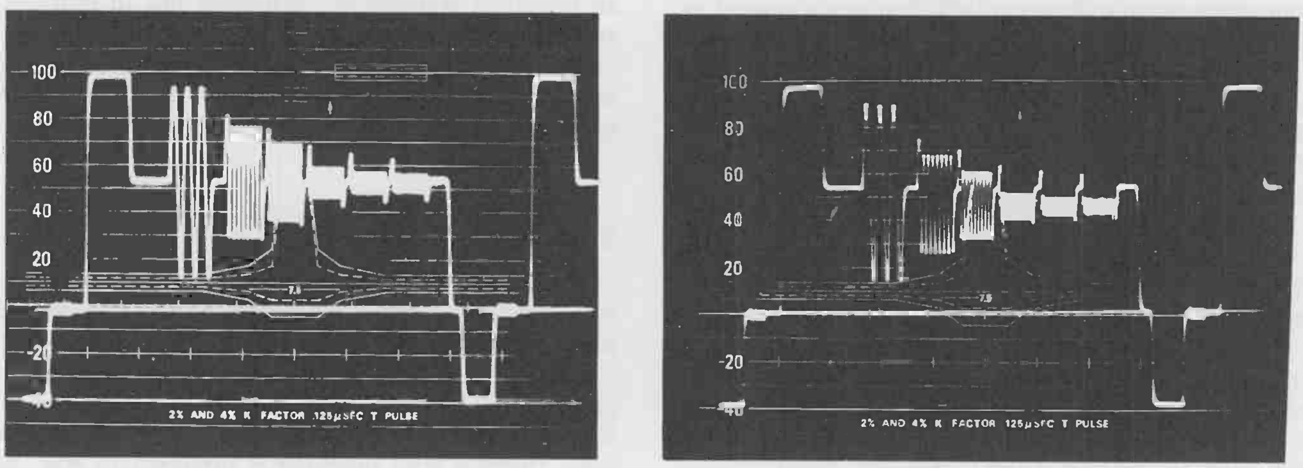
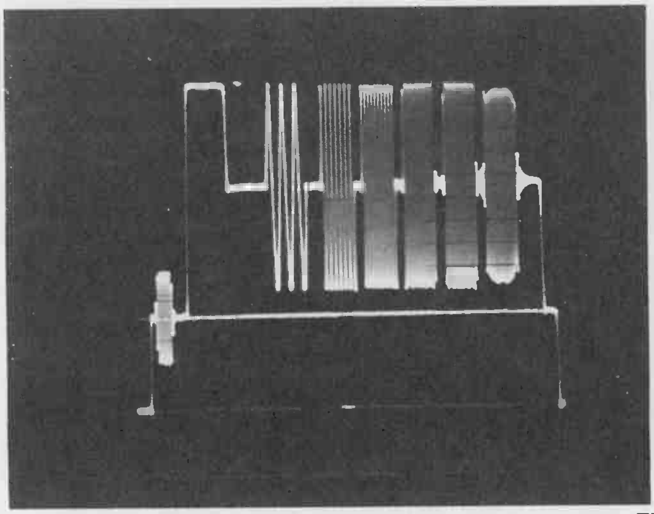


Figure 12.

Comparison between synchronous detection (on the left) and envelope detection (on the right) for a multiburst signal when the waveform monitor is operated with IRE roll-off. Note the axis shift (luminance error) in waveform on the right for burst signals above 0.5 MHz.



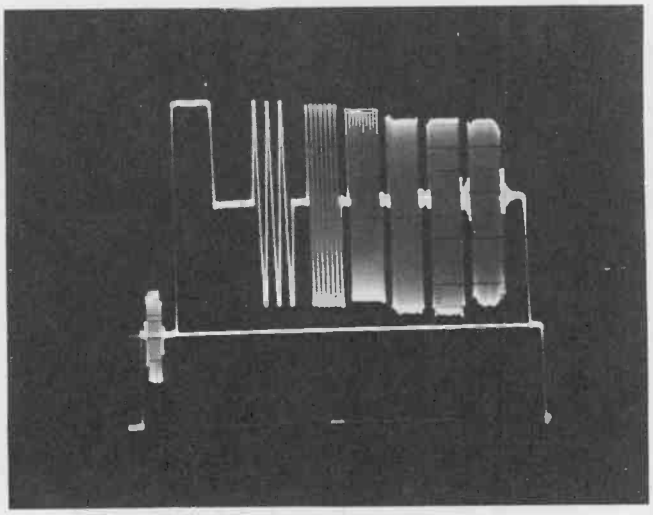


Figure 13.

Envelope detection of full amplitude multiburst signal when the IF is flat and bandwidth limited to approximately 5 MHz. In the *left hand photo*, the video amplifier is quite wide-band and phase-linear. In the *right hand photo*, the video amplifier is bandwidth-limited to approximately 5.5 MHz.

Most other test waveforms will also superficially appear to be correct in the wideband envelope-detection mode.

When the IF is not flat, as in the monitoring mode, then it is hard to say what video response is the best compromise for the most nearly correct reproduction of the many test waveforms in common use.

In one sense, the approach adopted simply hides the defects of envelope detection and has the disadvantage that waveforms seen on the waveform monitor may not correlate exactly with luminance values on the picture monitor.

The reason is that frequencies beyond 4.2 MHz which are being added by quadrature distortion and manipulated in the demodulator to make the waveform look good are largely ignored in the picture monitor due to its modulation transfer function (MTF) which is related to the kinescope spot size.

Consideration was given to limiting the video response to 4.2 MHz but the task force judged that the resulting test waveform distortions would be too difficult to interpret in normal operation in terms of proper performance of the transmission system.

Figure 13 compares a multiburst signal for the case of a wideband phase-linear video amplifier (left hand photo) and for a phase-linear video amplifier bandwidth limited to 5.5 MHz (right hand photo).

Envelope detection is employed. For the bandwidth limited case, the loss of 3rd harmonic of the 2 MHz burst reduces its peak-to-peak amplitude and, additionally, the loss of 2nd harmonic of the 3 MHz, 3.58 MHz and 4.18 MHz bursts produces axis shift (luminance error).

It seemed better from an operations point of view to have similarity between transmitter input and output waveforms when the transmitter is performing correctly.

In the end, no definitive video response standard was selected. Instead, guidelines were included to state what overall demodulator performance is to be achieved by tailoring the video response.

Synchronous Detector . . .

From the discussion thus far, it is apparent that picture impairments and waveform distortions generated in the demodulator in principle could almost completely be eliminated with synchronous detection.

Then a transmitter video input signal, bandwidth limited to 4.2 MHz, should look exactly like the transmitter output signal, except for degradations caused by an imperfect transmitter. Achieving this in practice is another matter.

First of all the transmitter being monitored must be free of incidental phase modulation—that is: AM to PM conversion. This will be discussed at a later point.

Phase Accuracy . . .

The next consideration is that the reference carrier in the synchronous demodulator must be very precisely held in phase with the transmitter carrier.[10]

This is not easy since the transmitter carrier normally is sampled only for a few microseconds at sync level or, preferably, at black level on the back porch of horizontal sync. Pedestal level sampling is preferred to sync level

10. **Werner Strossenreuther, "The synchronous detector in Nyquist test demodulators and TV monitoring receivers," Rundfunk Technische Mitteilungen, Jahrg 19, 1975 H.3.**

sampling in case the transmitter has some carrier phase shift between pedestal level and sync level.

If the phase lock is not precise, waveform distortion will be generated that looks much like low frequency envelope delay distortion at the transmitter. In fact, such demodulator distortion can be partially offset by adjustment of the transmitter envelope delay, but then the transmitter would be broadcasting an incorrect waveform.

Figure 14 shows a pulse and bar waveform at the demodulator output for 0°, 3°, and 10° phase error of the synchronous detector reference carrier. The committee Task Force selected 3° for demodulator performance tolerance.

Spectral Purity . . .

The demodulator reference carrier also must have a high degree of spectral purity in order that the demodulated signal be noise free.

Phase jitter of the reference carrier will convert to noise in the output signal by slope detection in the demodulator (PM to AM conversion). Standards were set for the reference carrier spectral purity.

Recommended Practices for Employing Standard Demodulators . . .

In order to assure uniform high-quality television signals transmitted from all television broadcast stations, it is desirable not only to have standards of transmitter performance and demodulator performance, but also to recommend which demodulator modes shall be employed during specific transmitter tests and adjustments, and for monitoring.

Otherwise, one station making adjustments using, for example, a wide-band envelope detection demodulator, might arrive at different on-air signal characteristics than another station using a bandwidth limited (sound notch in) demodulator with synchronous detection.

Appropriate recommendations for demodulator use are made in the standard but the details will not be repeated here.

Measurement and Monitoring . . .

If it were not for almost omnipresent incidental phase modulation on broadcast television transmitters, demodulator synchronous detection modes might be exclusively employed.

However, a transmitter with appreciable incidental phase modulation, adjusted using a synchronous detection demodulator, would produce some minor signal distortions on TV receivers using envelope detectors; and envelope detection is employed in the vast majority of receivers in use today.[11]

11. W. L. Behrend, "Effects of incidental phase modulation of TV transmitters, or other circuits, on TV signals," IEEE Transactions on Broadcasting, Vol. BC-19, No. 3, Sept. 1973.

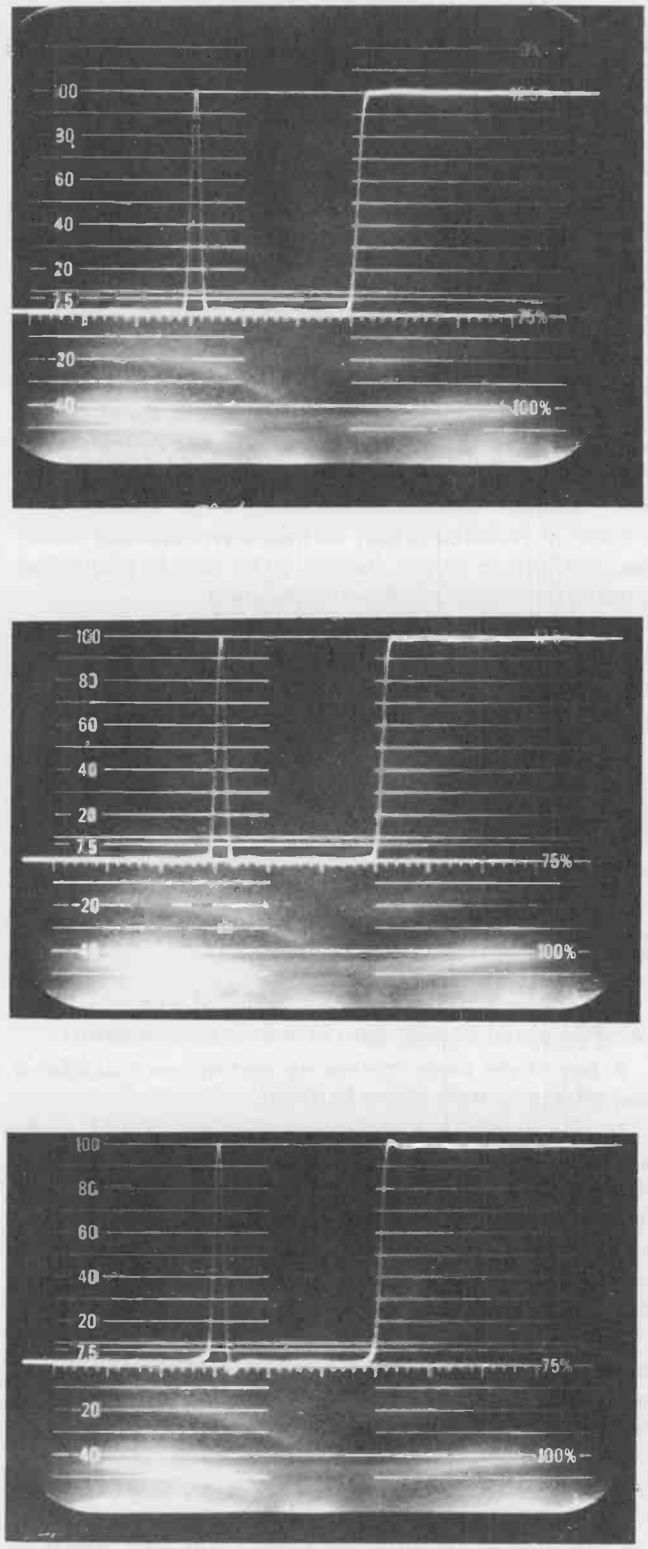


Figure 14.
Reproduction of 2T since sine-squared pulse and bar signal when the demodulator has 0° (upper trace), 3° (middle trace), and 10° (lower trace) phase error of the synchronous detector reference carrier.

The transmitter low frequency envelope delay correction and the differential phase correction, among other things, would not be correct for these receivers. No doubt the results would be acceptable in the preponderent number of cases, but transmitter adjustment would not be truly optimized for most receivers.

A guess is that most well-adjusted VHF-TV transmitters have incidental phase modulation in the range of 4°-8°, with somewhat higher values for UHF transmitters. Some out-of-adjustment older transmitters may have still higher levels of phase modulation.

The situation calls for care in interpreting the results of transmitter measurement and monitoring. It is hoped that the recommended practices detailed in the standard will be of help in identifying the presence of incidental phasę modulations and enjoying the demodulator correctly for transmitter adjustments.

Some demodulators may include a synchronous detection quadrature output channel to be used in evaluating transmitter incidental phase modulation.

Performance tolerances on reference carrier phase accuracy in the quadrature output mode are at least as stringent as for the normal mode, previously described, if transmitter phase modulation is to be reliably measured.

In addition, the 90° phase relation between the normal synchronous detection mode and the quadrature mode is quite critical.

Summary and Conclusions . . .

This paper has outlined the contents of a new standard being prepared for television broadcast demodulators.

A few of the more interesting performance standards and tolerances were shown in detail.

It was noted that each manufacturer should make available to the user, illustrations of commonly used test signal waveforms as they normally appear at the demodulator output. This will enable the user to distinguish between the effects of envelope detection quadrature distortion in the demodulator and transmission system imperfections.

More important to this paper than the standards themselves are the reasons behind the selected values and the compromises reached by the Task Force in arriving at these values.

This background should be of assistance in interpreting the standards as outlined in this paper and in providing additional understanding of the entwining transmitter and receiver performance factors and their effects as seen in the final picture.

Finally, the reader is cautioned not to rely upon, nor to invoke at this time, the performance standards and tolerances described in this paper and its illustrations.

These are subject to change during the review and approval cycle of the draft standard.

Acknowledgements . . .

It was indeed a great pleasure to work with all members of the Task Force, listed below, in drafting the proposed standard, and to share in their broad and diverse knowledge.

Each brought some specialized information, as well as overall knowledge, to bear on the work at hand.

Without reservation, it can be said that the entire Task Force worked together in a most professional way to create the best standard possible with no bias toward the interests of any company or organization.

The Task force members:
Mr. Gluyas, *Chairman*.
Alex B. Best, Scientific-Atlanta, Inc.
A Hans Bott, Harris Corp., Broadcast Products division.
Dennis Evans, Suburban Cablevision.
C. Bailey Neal, GTE Sylvania, Entertainment Products, Division, *Chairman*, parent BTS Committee.
Alex Kwartiroff, Telemet Division, Goetel, Inc.
Arne A. Lassila, Columbia Broadcasting System.
Ralph L. Mlaska, Public Broadcasting System (with WTOP-TV, Washington during most of his Task Force service).
Ulrich R. Rohde, Rohde & Schwarz.
Steven Roth, Tektronix, Inc.
Jouke N. Rypkema, Zenith Radio Corp.

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Panel Discussions

Technical Papers

Workshops

FCC Technical Panel

Moderator:

Joseph P. Gill, Jr.
Director of Engineering
Cosmos Broadcasting Corporation
New Orleans, LA

FCC Participants:

Wallace E. Johnson
Chief
Broadcast Bureau

Phyll Horne
Chief
Field Operations Bureau

Charles A. Higginbotham
Chief
Safety and Special Radio Services Bureau

Neal K. McNaughten
Assistant Chief
Broadcast Bureau

Dennis Williams
Chief
Aural Existing Facilities Branch
Broadcast Bureau

Mr. Gill:
We would like to start the FCC panel today by giving our panelists a minute or so to tell us what they've been working on, or just anything.

Mr. Johnson:
There are a lot of things that are happening as far as broadcasting is concerned.

First, re-regulation. I guess one of the questions I've been asked over and over again is: "What happens to re-regulation when Chairman Wiley leaves?" As far as the staff is concerned, we're dedicated to the re-regulatory effort, and we still need your assistance and help.

The Re-regulation Task Force will continue and we will do everything possible to try to simplify the rules so you can conduct business in as simple and effective a way possible, and that we can regulate in a manner which still meets the requirements of a regulatory agency.

We've got a couple of things coming up: the curtailment proceedings; the World Radio Conference (WARC); the possibility of AM stero; FM quadraphonic, and roadside radio, just to name a few.

Mr. Horne:
The Field Bureau has the responsibility of enforcing the Commission's rules.

One of the biggest problems we have is associated with the Re-regulation Task Force, which continually keeps changing the rules. From week to week, we're not quite sure which rule we are supposed to be enforcing.

To stimulate a few questions I'd like just to mention a couple of things that are of particular interest to the Field Bureau.

With the publication of a new Broadcast Operator Handbook, and through the efforts of Bill Heisinger, we've had very few Third-class examination complaints, and I hope that continues.

As you probably know, we have in the process a Notice of Inquiry involving operator requirements. We're now analyzing some 1100 pages that were filed in that proceeding. I'm not able to tell you at this time exactly where we're going, but we have promised Chairman Wiley that we will get something to the Commission by the middle of June. That doesn't mean that final action will be taken, but at least we'll be moving forward.

We have prepared a TVI handbook which we expect to have available from the Government Printing Office rather soon. It is designed to help the average citizen take care of his interference problems.

We have been infested with Channels 2 and 5 CB interference problems. Now, with 40 CB channels, Channel 6 is being added to the list.

A few other things:

Often we're asked, "Could we have some type of checklist with regards to your radio inspectors? They have a checklist when they come into our station that they follow, and we'd like to have a copy of it." We have now prepared copies of this checklist which will be available to anyone.

We have also prepared a booklet listing the most frequently violated rules. Again, in the matter of preventive maintenance, with a little care and attention the licensee can easily avoid those violations.

Mr. Higginbotham:

To tell you a little bit about the Safety Bureau, it's the oldest radio service. Included in the Safety and Special Radio Services Bureau are maritime mobile aviation, land mobile, amateur and, of course the personal radio services.

Mr. McNaughten:

For the past year and a half I have had to learn to speak satellite on behalf of the Commission and the United States. I just completed work with the World Administrative Radio Conference in Geneva, which was held in January and February of this year. We are now preparing for the 1979 World Administrative Radio Conference.

In that regard I would like to urge some of you to respond to the next two Notices of Inquiry which are coming out. If you don't, we're going to lose some of the AM, FM, TV and auxiliary bands to other services. There's also an awful lot of demand for the 13 gigahertz and the 6.8s bands which we must preserve for ENG.

Mr. Williams:

The Aural Existing Facilities Branch is responsible for program test authorizations—that is, licenses covering construction permits.

We also process minor change applications for AM and FM stations. We process Form #302 for the direct measurement of power, including those for directional AM stations.

We are responsible for pre-sunrise authorizations. And we oversee the installation of antenna monitors, and oversee the authorization of approved sampling systems and SCAs for FM.

We also process automatic transmission systems (ATS) applications.

We've made some progress this year by simplifying two areas relating to directional AM broadcasters. Such stations are no longer required to request special temporary authority every time they want to switch to nighttime pattern during the daytime hours to make antenna or monitoring point measurements. We've now authorized permanent authorizations to do that. Once you have such authorization in your possession, you can switch over at your convenience for taking measurements.

Secondly, we've reduced the requirements for remote control authorization for directional stations by having a simple form and excluding the one-year stability requirement for those stations that have installed the approved sample systems.

Question:

In taking an audio proof, stations using passive equalizers on the program line set them all back to zero to pass proof, then turn them all back up to 20db peaking in the middle and cutting everything below 250 and above 3000, and broadcast that way.

What is the Commission's position on the use of passive equalizers on the program line with regard to audio proofs?

Panel Member:

At the present time, we do not allow equalizers to be flattened out so that the proof can be taken. However, I would like to point out that we expect Rule Making to be initiated in this entire field sometime this summer to clear up problems that have arisen in the past few years that weren't around when the rules were written.

Problems such as equalizers, compression, limiters, FM stereos, and others that were never addressed when the present rules were instituted. We'd like to see a Rule Making where everyone could bring us up to date on it.

•

Question:

Under existing ATS rules, is it permissible to establish a second ATS alarm point, for example, in our program director's house—so we could operate with nobody at the station, but the alarms will ring in his house if something happens?

Panel Member:

The licensee may apply for an ATS monitoring point position other than the remote control position or the studio.

•

Question:

Would you briefly review the present controversy with the FAA concerning interference to navigational aids by FM stations?

Mr. Horne:

I presume you're referring to the recent situation where the FAA is objecting to FM grants on grounds of electromagnetic incompatibility between the station and the FAA facility.

We're still in the process of trying to resolve this matter. So far, we haven't got a decision as to how to follow through on this.

The FAA has not come up with any kind of standards—which has us greatly disturbed. We've had a lot of discussions with them trying to determine what limits they will tolerate, but we haven't been able to pin them down on that. They're mainly concerned about potential interference problems which may cause a safety problem.

Question:

Will the present definition of modulation that are contained in the ATS rules be universally applied to all stations?

Panel Member:

The 10-in-one-minute definition for the measurement of modulation now contained in the ATS rules will be applied uniformly.

•

Question:

Under ATS rules, the modulation sampling circuit is supposed to be at the transmitter output terminals. Is that also the place now to do a proof of performance, if that's where you're determining your modulation percentage?

Panel Member:

Yes.

Questioner:

Rather than something off the air?

Panel Member:

Either way is acceptable.

•

Question:

Can we look forward to pre-sunrise operation on Canadian clear channels?

Mr. Johnson:

As far as pre-sunrise is concerned, we've had two meetings with the Canadians to discuss various problems and pre-sunrise is one of the issues we are discussing with them.

So far, I don't see any indication that they are going to relent in any way on the amount of protection they want. So at this point, I wouldn't promise that we're going to get anything more as far as pre-sunrise is conerned from the Canadians. But at least we are discussing it with them again.

•

Question:

I'd like to enlarge on that previous question about the output from the transmitter being used for the modulation monitor indication and proof of performance. If that's acceptable, why isn't the rule changed so that we don't have to go through all of the circuits, including the antenna circuit, for proof of performance?

There are very few licensees in the country that have ever made a proof of performance through the antenna circuit by taking their sample from a signal that has been radiated from the antenna. It's always taken off the transmitter output. Yet, the rule makes instant criminals of all of us by telling us that we've got to go through the antenna circuit. Nobody ever has.

If that's the case, why can't we make a proof of performance into a dummy antenna, to reduce interference where there is 24-hour-a-day station on the channel?

Panel Member:

As a matter of policy, we have allowed a station with a 24-hour operation and has an auxiliary transmitter to switch on the main antenna while taking its proof. We have permitted that in the past.

The point is the antenna must be connected during it. The proof can't be done into a dummy load, with the exception I've just mentioned, which is not applicable to most stations.

Questioner:

You're forgetting the hundreds of daytime-only stations that cannot operate at night on Class 1A channels because there are stations already operating with regular program service on a 24-hour basis. What are those stations going to do?

They can't radiate a signal and make a proof. Are they going to violate the rule that says we cannot interfere with regular program operations? They've got to make their proof on a dummy antenna.

When a transmitter manufacturer makes type-acceptance performance measurements for their equipment, they're not radiating on the air. They're radiating into a dummy antenna. That's the way the certification is made in the first place.

So why does the licensee have to conduct his proof through the antenna system?

Panel Member:

You can operate during the experimental period to take the audio proof.

Questioner:

No, you can't, not without violating the rule that says you can't interfere with stations that are on regular program service during the experimental period.

Panel Member:

Did you receive a specific interference complaint from the dominant station?

Questioner:

No, but I'd really like an answer to this.

Mr. Johnson:

I think this is something we should look into, as far as our re-regulation is concerned. As far as I know, we don't have any special request to conduct the proof into a dummy load. But I think that's something we should do.

Question:

Did I understand correctly that a station that normally operates 24 hours a day is permitted to do a proof on the dummy load while operating with the standby transmitter?

Panel Member:

Do you have an auxiliary?

Questioner:

We have an auxiliary transmitter and a standby control room. But we've been shutting down at least once a year to do a proof.

Panel Member:

We have permitted that to be done on a dummy load, in the past, if you didn't have a regular maintenance night in your schedule. If you have a normal maintenance night, we expect the proof to be conducted at that time through the antenna system.

We handle this problem on a case-by-case basis, by specific request from the licensee.

Question:

Who should be contacted for authorization?

Panel Member:

A telegram to the Broadcast Bureau requesting permission to use a dummy load.

•

Question:

Can anyone give me some idea what sort of timetable the Commission is looking at on the granting of AM applications that were received before the freeze?

Mr. Johnson:

Needless to say we got many more applications than we'd ever bargained for, and I can assure you that nobody at the Commission, in the foreseeable future, is ever going to recommend another freeze.

We're in a jam as far as AM applications are concerned. We've hired four new engineers. We've got a big push from our computer program, and we're hoping that by this fall we'll be in real good shape again as far as AM processing is concerned.

Questioner:

At the present time, we own and operate five TV translator stations and have applications pending for several more. We have just finished installing our first FM translator, and have run into a problem regarding operator requirements and the logging every six hours between 8:00 a.m. and 10:00 p.m. To comply with the rule we had to install a phone line between the FM translator and the control point at a cost of a couple of hundred dollars.

Can we get some relief from these requirements? We can operate a 1000 watt television translator without such problems but a one-watt FM translator requires a $200 a month extra cost to comply with the rules.

We are using an off-the-air receiver tuned to the translator and feeding the signal back on a phone line to the TV and AM control room, so we can determine what's happening there. The problem is the Communications Act, as amended, exempts TV translators from the operator requirement, but nobody thought to include FM translators.

I believe the NCTA filed a petition to block any Rule Making that might change this. Am I not right?

Mr. Johnson:

There are about 12 petitions pending as far as translators are concerned. One of them was filled by the NCTA where they're asking us to more clearly identify the policy that we're going to follow in the future as far as translators are concerned.

Their viewpoint is that translators are competing with cable in an unfair way. They want us to take a look at the situation and determine what role we think translators should play in the future, taking into account the problems that we have.

Questioner:

We've run into that on one of the television translator applications where we're trying to improve the signal in our Grade A contour. They say we were doing them a disservice by even requesting the translator application in that area.

I was just curious about the FM. The one-watt FM translator's have caused us more trouble than all of the television applications we've filed.

Mr. Johnson:

We've granted very few FM translators, and I guess we're not cognizant of all the problems that you've brought to our attention.

•

Question:

Some of us have been waiting for about a year on a Rule Making authorizing circular polarization for television. I'm wondering if you could comment on what the status of the Rule Making is and when we might expect a decision.

Panel Member:

Circular polarization for television will be on the Commission's agenda on April 7th.

•

Question:

How does the FCC handle a CB interference complaint from an irate viewer? And what is the broadcaster to do when this irate viewer then calls up the television station and says, "Hey, I can't get anything done?"

Mr. Higginbotham:

We have so many TVI complaints that involve CBers and others that it's swamped our field offices. I think we've had about 100,000 or so complaints in the last year. We are trying to change all of that.

Mr. Horne has prepared a new TVI bulletin. We've also been working with the Advisory Committee which was formed expressly for the purpose of addressing some of the pressing CB problems.

There are a number of things we feel have to be done. One of them is that the CB industry has got to go to work educating the serviceman. We have a lot of complaints of interference, but few people out there are qualified or well-informed to deal with them.

We can't tell a CBer, "Go get your set checked," when the CB serviceman has very little awareness of the things he should be looking for in the case of TVI.

The TV serviceman has the same problem. He thinks that the TV sets that he services are not susceptible to CB interference, and he doesn't really know what to do to correct it.

We feel the first step that must be taken is to get some kind of an educational program underway and the Advisory Committee is working on that.

In the case of the broadcast industry, remember that the average viewer, as well as the CBer, are not technical people and they don't understand the problem. I think somewhere along the line we have to find a way of getting that message to the viewer, through some type of public affairs programming. Give some consideration to spending a few minutes, sometime during the year, to explaining what to look for in terms of interference.

Would Mr. Horne like to add anything to that?

Mr. Horne:

We don't have the personnel to investigate each interference problem and generally our advice is to send out a primer giving the viewer counsel on what the problem is so he knows what to do.

We ask for identification of the CBer, because we can't go out and locate him. We then write to the CBer and ask him to have his equipment certified to see that he is operating legally.

Once he's operating legally then it's up to the recipient of the interference to take care of it.

Also, we get complaints, particularly with regard to interference to stereos, organs, public address systems, etc. As far as we're concerned, that's their problem. They receiving signals they are not supposed to be receiving, and they've got to take care of it.

•

Question:

Do you think we might get a decision on the clear channel breakdown in our lifetime?

Mr. Johnson:

We have already started our work on the clear channel proceeding, but it's just anybody's guess as to when something will be coming out. We've gone through the comments and we're trying to organize our efforts to come up with something as far as the Commission is concerned.

There are some other things that are going on, other than the clear channel proceedings, which you may be interested in—such as the World Radio Conference.

In that proceeding there's one possibility of extending the AM band, both above and below 540 and 1600, to get some additional channels above and below the present band.

As far as the clear channels are concerned, it'll be a very hotly contested case, and it may be some time down the pike before we finally come to a conclusion.

•

Question:

I guess 90 per cent of the broadcasters in the country use the indirect method of FM power measurement due to the expense of trying to use a dummy load and a calibrated watt meter to come up with their actual power. From the ATS standpoint, would it actually be more accurate to use some type of line sample to see what the transmitter's putting out, rather than going by the plate current and plate voltage method?

Is there a system where you could actually calibrate your reflectometer using the indirect method and then use that as your indication of power output?

Also, has any thought been given to being able to make the impedance measurement on FM antennas once and get it certified for direct method, the same way as an AM antenna? There are some AM antennas that haven't been measured in 20 years, and still operate using the direct method of power determination.

Panel Member:

When we were working on the ATS rules our guiding philosophy was not to try to come up with specific rules that would apply only to ATS stations, but, rather, just deal with the ATS problem itself, and what specifications and what the general rules would be.

Now, the direct method of calibration regardless of whether they're ATS or not should be made in a separate Rule Making proceeding. We didn't want to just carve out a special relaxation for ATS stations as far as the technical standard.

We'd want to deal with that in a separate Rule Making. And the same would be true of the FM antenna problem that you mentioned, about making measurements and then having some sort of long-term direct method. We didn't want to handle that in the ATS ruling. We wanted to keep that separate.

Questioner:

I wasn't thinking of it from just ATS, but it's something that came to light with ATS, and maybe it would be a benefit to put all broadcasters in step with today's technology. Since you can accurately measure the antennas, I think when the original direct/indirect rule came out, the equipment was not that accurate and available to actually make good measurements on antennas at that high a frequency.

Mr. Johnson:

What we'd appreciate is that whatever data you have on this subject be sent to us, either in the form of a Rule Making or just informally—in letter form—and let us take a look at it and see what we could do.

•

Question:

We had people come to us to have a three-inch line section and meter calibrated to measure power by the direct method. I assume that they had gotten some kind of authority from the FCC to measure power this way—with the stipulation that the unit had to be recalibrated every six months. This seems like a good way of measuring FM power output. But if you take an RF sample, it's a lot more accurate.

I just wonder where this ruling came from; whether that is actually the way the FCC has spelled it out, and how would we go about getting a relaxation on this six-months rule?

Mr. McNaughten:

The rule has been there for ages and technology has overtaken it. The time is right to deal with this problem in a separate Rule Making.

The point I was trying to make is we didn't want to confuse the ATS Rule Making with those separate concepts that would apply to all FM stations, regardless of ATS. We would welcome whatever you have on that, either in the form of a petition for Rule Making or informally.

•

Question:

You mentioned that you didn't want to make special rules for ATS operation although the rules you adopted for modulation control, the definition of peaks of frequent occurrence seems to be a very special rule made for ATS.

You defined the window as being five milliseconds, and the broadcasters has no convenient way to check this. A normal monitor, say for AM, will have a peak light that operates a duration of 150 to 300 milliseconds. The stereo monitor rules specify that the peak light must stay on the three seconds.

So, with this very special rule of a five-millisecond window, the broadcaster has no way to correlate the information from a modulation controller to his monitoring devices.

Secondly, five milliseconds is a very short duration. If you take the average overmodulation peak, it probably would have a duration of about maybe 10 to 15 of these five-millisecond peaks. So that in reality, one overmodulation peak would occupy enough five-millisecond bits to count as excessive modulation in one minute.

I don't think that is a very realistic rule, and should be considered in one minute.

Another one is that you specify one overmod peak at 125, then overmod peaks at 100% negative. Well, your 125% peak doesn't do any damage other than exercise the transmitter. The 100% negative peak does all the damage. So if you want to limit something more than the other, then it should be the other way around.

Now, I do have to compliment you on one thing: that you did come forward and make a definition of excessive overmodulation. Right or wrong, it is a definition. So now we can all join together. It is something that is workable.

Mr. McNaughten:

You're correct in what you say. I don't believe we intended to carve out a special case for ATS. It's simply that we had to come up with something that deals with a machine, what a machine can do, for the first time, versus the present rules where you have an operator.

The ATS Rule Making will be an on-going thing. We still have the directional AM and TV stations to address. So I think any comments that you would have along those lines would be helpful.

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Question:

It is my understanding that irrespective of what the rules say about a calimetric load for calibrating the power meter on an FM transmitter, that the Broadcast Bureau does accept as proper a direct-reading watt meter that is not calimetric, but that is calibrated. I'm talking about an air-cooled load that is calibrated by the manufacturer every six months as being satisfactory for direct measurement of power.

Mr. Williams:

That would be satisfactory as determining the power into the dummy load, and then that used as your standard for calibrating your line meter.

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Question:

Although it's fairly recognized that you should have in your control room the capability of actually monitoring your on-air status, is it a rule that the operator who is actually in control at the time have access to, or have in operation, an on-air monitor in the control room?

Mr. Horne:

Have access to it.

Questioner:

Should it be on, should he be monitoring on-air, or can he select an actual off-air in-house monitor? In other words, say he was on the air at three in the morning, and, he's the operator in charge of all transmissions, is there a rule that he must have an on-air monitoring system?

Mr. Horne:

He should have. This again will give you a little latitude.

We're trying to be lenient with the broadcasters and to cooperate. In this operator-on-duty, we had quite a go-around here a couple of years ago on the matter of his position — he had to be there so he could look at the meters all the time. That gets a little impractical.

It's a matter of his being available and watching it when he's on duty, so that he *has the capability of observing it* continually. But he doesn't have to sit there and watch it continually.

Questioner:

Then you could actually select either monitoring system.

Mr. Horne:

You could, either one.

Mr. Gill:

Thank you very much, gentlemen, for a most informative and interesting hour.

The Utilization and Application of Character Generators Panel

Plus Questions and Answers

Moderator:

Robert J. Butler
Director, Technical Division
NBC Television Network
New York, N.Y.

Participants:

Frank D'Ascenzo
Project Manager Video Products
3M Company
St. Paul, Minn.

Thomas Hindle
South-East Sales Manager
Thomson-CSF Laboratories, Inc.
Stamford, Conn.

Eugene Leonard
President
Systems Resources Corporation
Plainview, N.Y.

Thomas Meyer
Product Specialist
TeleMation, Inc.
Salt Lake City, Utah

Mr. Butler:

Good morning ladies and gentlemen. It is my pleasure to serve as moderator for today's panel, "The Utilization and Application of Character Generators."

We have with us this morning a distinguished group of experts who specialize in the field of character generators. These are the gentlemen who make the units that we buy and use in our studios.

Let's just talk a minute about what a character generator is. We've all seen them and used them. Unlike a camera, or video tape machine, there are no measurements that we can rate character generators on, such as differential phase differential gain, signal-to-noise. These analog specifications are meaningless when you talk about a character generator since it is a digital device. The measure of how well a character generator performs is how did it look and how easy is it for you to make it look the way you want it to look.

Hopefully, at the end of this panel discussion, you will have a yardstick by which you can measure the performance of the equipment shown here at the NAB and in different sales presentations.

In order to get to the question and answer session as soon as possible, we've chosen the following format. I will call on each of our panel members to give an overview of what their product might offer you. After that, each gentleman will have five minutes to give his opinions. We will then be ready for questions. First, I'd like to call on Mr. D'Ascenzo.

Mr. D'Ascenzo:

Perhaps the first (or at least a very early) application of an electronic character generator in a broadcast television facility took place in late 1967, at Channel 2 in Dade County (Miami), Florida. Using what was then a prototype of the later-to-be-introduced A. B. Dick Model 990 Character Generator, Channel 2 succeeded in video taping eight, one-half hour segments of a complex in-service video training series in the surprisingly short time of two working days.

What was significant about this was the fact that over 1,200 "supers" were used in the eight shows. That's an average of 150 per show, or about 5 per minute. Imagine the problem of handling over 5 super cards per minute; simulating vertical and horizontal "reveals"; effecting transposition of words; and doing all this on cue with a rather technical script. Many long hours had gone into artwork preparation, rehearsing, and taping prior to the arrival of the Model 990 — all to no avail.

The success achieved with this early prototype unit paved the way for the subsequent entry of the A.B. Dick Model 990 Character Generator into the broadcast television industry and the eventual evolution of what has become one of the more practical product developments of the '70s.

A number of exciting developments in character generator evolution have taken place in the 10-year span from 1967 to 1977. So many, in fact, that character generators are no longer thought of as simple titling machines — i.e., an easier, better way to create a television super — but as true graphics tools.

From the standard dot-matrix character of 1967, modern day machines have advanced to presentation of truly high-resolution alpha/numeric displays and multi-front capability.

Applications of video character generators in commercial and educational television stations are now widespread, and examples of usage are seen every day on local and network shows. In addition to general titling, they are used to produce show credits, store and present statistics (for example, a baseball player's batting average), display election returns, superimpose news flashes over regular shows, and for game show graphics, advertising messages, and a variety of other purposes.

At first, the application of a character generator was quite limited. Pure and simple name supers and perhaps credits. Early machines ran headlong into a problem, however, because of their rather poor character formation or resolution capability. The demand of the professionally-oriented TV industry forced manufacturers back to the bench in an effort to develop systems capable of emulating graphics-quality type. And once that was accomplished, the industry demanded more. Happy to see high-quality, electronically-generated characters, their demand grew to include multi-front capability in various sizes.

The circle, of course, never closes.

Some Background . . .

Tracing the technological history of titling systems begins when DTL was a rather new technology and core memory was still the only way to go.

An early titling machine (i.e., 1967-1970) was usually restricted to one keyboard input and one video output. Bulk external memory systems were rudimentary at best, with one or two tape-type and hard-disk devices at the forefront. Internal memory was a major shortcoming in the design of these early systems. Solid-state RAM (Random-Access Memory) technology was in its infancy, and expensive as well. Core memory and then-current static and dynamic shift registers were in common use.

During the ensuing years of 1971 through 1975, the major technological improvements in titling systems were naturally tied to integrated circuit improvements. The ready availability of solid-state RAM's. 16k-ROM's and various other sized ROM's and PROM's opened up new horizons to designers. Multiple entry, dual channel video output, larger in-machine data (page) storage capability all became realities. Even so, the actual system structure of equipment did not change drastically.

Along in the early '70s, the idea and then the realization of "software" oriented titling systems emerged in the form of an advanced electronic titling device and changed the thrust of all future design thinking and customer expectations. Even then, memory was still a major problem, since lower-cost IC-type memory devices were not yet readily available. While impressive in capability (at that time), the equipment was relatively expensive. While the video output capability was still limited to a single channel, multiple inputs are readily accommodated.

In 1976, industry began to realize the promise and power of the newly available microprocessors. These computer-in-a-chip devices have been hailed as the catalyst for a revolution in electronic equipment design at all levels, and a simple observation of what is happening in the area of video games and hand-held calculators in a sufficient example of the "revolution."

An Innovation . . .

Microprocessors have altered the basic structure of titling systems as well. This can best be illustrated by describing in some detail a new product development which has evolved from the Datavision Laboratory of 3M Company.

The 3M-Datavision Model D-8800 ECG Titling System in a modern television titling and graphics system designed around microprocessor technology. The initials "ECG" stand for *E*lectronic *C*reated *G*raphics and underline the forward thinking features incorporated in the System. The D-8800 utilizes a multiprocessor design based on microprocessor technology and powerful operating programs to allow broad *creative* freedom and systems control.

The System design is based on the concept of maximizing software control to produce a SIMPLE TO OPERATE yet flexible and complete television graphics system. Software control provides the inherent capability to update the System over the years without incurring extensive hardware modification, And, special application or customer requirements can be more easily accommodated by software changes in the internal operating program.

The System is supported with an available library of 30 font styles in various sizes. Additional fonts will be added on a continuing basis, and custom design font service is available. Up to four complete fonts may be stored within the D-8800 for intermixed display. Additional fonts are stored on floppy discs, and may be quickly and easily loaded into the System.

Character resolution is defined in 35 nanosecond increments to produce characters of excellent smoothness, a far cry from the earlier stair-step character structure popular only a few years ago. All fonts are carefully designed to provide the best possible "video look," and characters are of variable width (proportionally spaced) to provide optimum aesthetic appearance on the TV screen.

The maximum character height is 256 TV scan lines. Minimum practical height is 16 scan lines, although smaller characters may be defined within the limitations of the system and, of course, readibility. Larger characters or logos can be displayed with special font design and multiple keystroke entry.

All graphics are composed at the Model D-8800 control console. The console features an **Inter-Active** panel display, a standard typewriter keyset, auxiliary keys for editing, message storage and retrieval, and system control purposes.

A variable number of characters may be entered on any given row. While the maximum possible number is 100, for normal display purposes the number of characters used per row usually would average about 20-32. The number of rows utilized per display is also variable up to a maximum of 16 rows. Again, character design and variety per display determines this limit.

Both messages and font information is stored on removable "floppy" diskettes. While the basic system is equipped with one disk drive, up to four may be easily accommodated. Typically, 20 or more full fonts (1,960 characters) may be stored on one deskette. The transfer of a new font from disk to the D-8800 occurs in less than 5.0 seconds.

Floppy disks are also utilized for message storage. In this case, each diskotte will handle up to 152,000 characters, or the equivalent of 6,000 25-character rows of title information. For composition ease, the 6,000-row capacity is formatted into 1,000 addressable pages, each page of variable size depending upon font style, size, and mixture. Messages are selected on a random basis via a numeric key pad on the control console, and playback occurs within one-half (0.5) second of selection. Each message may carry "preamble data" to define roll, crawl, mask, speed, position and direction.

One of the more outstanding features of the Model D-8800 is the display panel mounted above the control console. In conjunction with the operating program, the display panel provides operator support through a carefully designed "operator system" hand-shaking program.

The inter-active nature of the system helps anyone become a qualified operator in a minimum amount of time and removes the system from the status of "specialized" equipment to the "general use" category.

The Model D-8800 provides the usual dynamic features expected in a superior graphics/titling system:

- Vertical roll (6 rates)
- Horizontal crawl (10 rates)
- Character or word flash
- Character-by-character color encoding
- Roll, crawl, and pause
- Accurate centering

Additional standard features, however, include:

- Dual high resolution channels
- Vertical roll *up* and *down*
- Horizontal crawl *right* and *left*
- Programmable roll/crawl masking (8 positions)
- Dual channel mixing
- Animation mode

The model D-8800 is equipped with *two* high resolution video channels which may be used independently in a preview/program model: concurrently as separate program outputs, or *combined* as a single channel output.

When combined, the dual channel capability allows creation of unusual effects, such as roll down of information, followed by crawl right of different information. The two channels may also be used simultaneously as two different outputs, and one channel may be in a static or update mode while the other is in a dynamic roll or crawl mode.

Another creative feature, the Animation Model, allows preparation of unusual animated graphic effects through use of the high-speed playback capabilities of the disk storage system. Two animation rates (up to 5 or 10 pages/second), and a specially designed "line segment font", puts animation at the operator's fingertips.

Inside the D-8800 Dual Intel 8080 microprocessors operate in conjunction with special software programming to enable the multi-function capabilities of the System. Here, the microprocessor has made its mark and foretells the future of the industry.

Inherent in the design is a rather generous use of internal memory. At every stage of systems structure, memory is being utilized to buffer and store information and to provide the digital intelligence of character formation. An accounting of internal memory capacity would indicate the following statistics:

Function	Memory Bits
Font Memory (one)	128,000
Page Display	128,000
Working Memory (Display)	32,000
Program Memory (Display)	32,000
Program Memory (I/O)	32,000
Working Memory (I/O)	128,000
Total Internal Memory	480,000
Three Additional Fonts	384,000
Grand Total	864,000

This represents an impressive and powerful operating system which would otherwise be prohibitively expensive if it were not for the readily-available microcircuits of today.

But why use *dual* microprocessors? In the D-8800 System, the processing load is split between the two internal computers. One Intel 8080 is assigned to the disk operating system, keyboard control, and external data Input/Output functions, while the other takes care of roll and crawl dynamics, refresh of Channels 1 and 2, and editing functions.

Operating two 8080 devices allows a greater degree of design freedom and provides room for future system growth from a software point of view. Indeed, a system like the D-8800 promises a longer useful life than previously conceived systems *because* of its software-oriented nature.

Goals Accomplished . . .

At the inception of the D-8800, four goals were uppermost in our minds. These goals were to design a system that:

1. Provided the user with a modern machine utilizing the most advanced micro-logic components.
2. Provided advanced system features at a competitive price.
3. Provided inherent long-life usefulness through internal "operating program" update rather than extensive hardware modification.
4. Provided in an easy to use system even though complex in concept.

These goals have been accomplished in the Model D-8800 ECG Titling System.

While the D-8800 and its competitive counterparts are ushering in a new era of Electronic Titling Systems, it's well to recognize that technology advancements and customer demands are never-ending.

A natural extension of titling systems extends into the general area of Video Art. Already, tentative systems have been shown, and we can expect this Video Art concept to mature over the next several years until animated, user-composed Video Art Systems are real, on-the-market machines ready for use.

Indeed, the TV studio artist of tomorrow will need to learn to lay aside his ruler and brushes and take up the joystick and light-pen as digital electronics designers learn how to develop systems to make the *TOTAL* graphics-for-television job do-able in a television format.

Mr. Butler:

Thank you very much, Frank. Our next panelist is Thomas Hindle of Thomson-CSF Laboratories.

Mr. Hindle:

Thank you Bob. I'll make it very short.

The utilization and application of high technology equipment in the television medium must be advanced in a manner that maintains the balance between the performance, operation and technical demands of the medium. Particular consideration must be given to the variety of personnel skills that are required to meet these demands.

Changes in programming, production and other viewer visible styles and techniques that govern the success of a broadcaster takes place in an evolutionary manner. This close relationship between "on air" programming and "in house" operation gives us direction in the development of equipment that will operate successfully in this environment.

The television medium consists of visual stimuli which are most effectively utilized when created, changed and refined in the same dimensions that an audience sees the final presentation.

Traditional methods to create graphics can not be used in this type of direct operation, since there are several intermediate levels of preparation before final presentation can be put to the viewer.

In order to reduce this time gap and number of operations between the original graphic and its presentation, it must be generated in a form that provides direct operator manipulation while in its final presentation form.

The introduction of Vidifont in 1969 applied high technology developments to electronically generate quality graphics that were, until that time, only achieved from "hard copy" artwork. With the characters in this format, a reduction of the number of intermediate steps between initial and final presentation of the graphic is realized.

The integration of these two facilities, electronic generation and manipulation of graphics, provides an effective closed loop system that ties together man, methodology and medium.

This inter-relationship of immediate operator feedback must be maintained as continuing technological developments brings about more sophisticated forms of graphic transformations.

Mr. Butler:

Thank you Tom. Now, Eugene Leonard of System Resources.

Mr. Leonard:

The emphasis on digital technology in television, and the realization that the character generator is a forerunner of complex digital capabilities to come makes this panel presentation singularly appropriate.

The character generator represents the most common form of digital image creation and manipulation presently used in the TV broadcast field. It is the prime generator of still frame graphic imaging in flat color, as distinguished from camera-generated full gray scale, full color moving images.

The television still frame graphic area needs less memory and manipulation capability than full gray scale and color television. Therefore, it is the first to offer economically-feasible capabilities in the form of character titling and graphics generators. These are presently confined to still frame, but they will soon become available in animated graphics and — we can be sure — will eventually be available in full gray scale and color.

In this connection, a new area of the television discipline can be expected to open in which we deal with manipulation of the contents of the frame by digital technique. This will provide a new dimension to the capabilities presently offered by the application of a digital technique to editing of frame sequences, picture quality enhancement and time-basis correction.

What are the capabilities which are so desirable that broadcasters will spend money and character generators and their more sophisticated sisters — the graphics generators?

Clearly, television exists in two forms. One is entertainment; the other, information.

The entertainment form is a vehicle for offering information which the sponsors of commercials pay for in order to support the television system. (This is not to imply that commercials are devoid of entertainment value in themselves.) Even in the area of news and weather reporting, it is apparent that there is a great entertainment content as well as hard information.

The television broadcast stream is the equivalent of 80 million bits of information per second. This is a far greater amount than the human eyeball-to-brain system can possibly accommodate. However, it is necessary in order to duplicate the world as human beings normally perceive it.

Most of the information coming to the eye is never stored in the brain, but that huge collection of information is sorted and partially chosen so that we wind up with various significant emotional and "non-hard information" memories in the brain.

Hard information is necessary, however, in the television system — whether it is the display of a sponsor's telephone number or the identity of a news personage, or a clarification of weather projections in the form of maps and charts, or other descriptive and educational material.

All of these areas demand a much simpler form of information, and this is where graphics comes in.

For graphics to be cost-effective, they must be camera quality, they must be versatile, they must be easy to assemble, and they must be readily available to personnel trained in the graphic arts who understand the impact of a hard graphic presentation on the viewer's mind. How does this translate into machine characteristics?

The following are eight key items in the evaluation of machines intended to accomplish character generation, titling and, eventually, graphics tasks:

1. Starting point resolution.
2. Multiplicity of logos, symbols and type styles.
3. Ease of creating new logos, symbols and type styles.
4. Ease of composition.
5. Flexibility of control locations.
6. Security of recall.
7. Multiplicity of output channels.
8. Vector toward the future.

Starting point resolution is a key characteristic of a character generator.

The camera offers practically infinite starting point resolution in that the video signal can begin to rise at any point along the scan line.

In digital devices, this change in video can only occur at intervals determined by a "clock" ticking away as the scanning beam moves from left to right.

It is easier for the digital video generation to match the normal concept of resolution regarding minimum picture element width. With a 27-nanosecond clock, that time becomes the minimum picture element width — but it is so far beyond the ability of the scanning beam to respond as to be useless.

Multiplicity of logos, symbols and type styles can be keyed on the same display. Prior requirements indicated that six was an ample number. There have been requests to increase this number to 12.

There is no way in which to standardize the efforts of the typographical artist. It has been suggested that a standard known character, in a known type style and of a fixed height (for example, 32 TV lines), be adopted by the industry as a measure of the code compression effectiveness of a character generator. In other words, how much of the available font memory in the machine is required to hold the shape information of this standard character?

Perhaps some IEEE or SMPTE committee can begin to consider this question.

As the demands on the graphic equipment have become more complex and artistically exacting, the creation of logos, symbols and type styles has been troublesome for some manufacturers and users.

Where it was originally sufficient to simply put a character up against a background, shoot it with a camera and digitize the result, this is no longer sufficient. Now, much more sophisticated adjuncts are possible, where a character is enlarged on a grid and a pair of crosshairs is positioned for "trimmings" purposes. This speeds up the generation of new logos, symbols and type styles immeasurably. At the same time, quality is greatly improved. This device also allows shapes to be generated directly by the "trimming" process without the necessity of a camera.

Although composition does not have to be done under enormous time pressure, it is important that the graphic composer be minimally involved with keystroking and multiple operations to achieve a given effect.

The effects that are important are flexibility in types of edging, in the variation of background for different rows, in the ability to fine-tune character positioning after the message is composed, and to have characters which are distinctive, such as a totally black character or a black character with a white edge, or a see-through character in which only the edging appears to cut through the program material.

The Chyron Approach . . .

The **Chyron III,** while not offering the equivalent flexibility, has many features which are desirable in a titling generator.

Chyron's original character generators were visualized primarily from limited applications in master control and engineering. In the interim, users have developed so many other applications for the units that a totally new and flexible system for locating and allocating keyboards and control systems has been devised.

A most important concept is the ability to integrate the system with an external computer which is supplying data for events such as elections and sports operations. In this connection the Chyrons have been remarkably easy to interface, including the various control codes which must be transmitted in order to allow the computer to do all of the things it should do. This was elegantly demonstrated in the NBC coverage during Election night.

Although a number of the other devices on the market do not include the requisite capability, past experience in digital operations has drilled into us at Chyron that error-checking is required whenever digital information passes from one medium to another.

While it is usually satisfactory for the internal electronics to bypass error-checking, except for such simple checking as parity, the information that is transmitted from digital circuitry to magnetic disc surface must be secure. Similarly, protection of fonts against instantaneous power failure is important since a power surge may lead to garbage appearing on the screen at the moment which, by Murphy's law, will be least desirable.

It is a basic principle in digital technique to use only as much equipment capability as is actually required to do the task at hand. This has given rise to the concept of a hierarchy of memory and capability in all digital systems operating in the real world.

At Chyron, the first requisite was the ability to recall information that had been previously stored in message memory, update it, and return it to message memory. This "preview and edit" function was seen only by the machine operator, and therefore could not justify a full-fledged font generation. For these purposes, a stylized preview display was adequate, provided the operator knew what font and what color was being used for each character on the screen. This information is provided by the diagnostic prompting display on the bottom line of the television monitor.

As complexity grew and other functions began to appear, it became advantageous to add a full quality second channel.

Chyron technology and system philosophy indicates that this should be a "frame grabbing" second channel which can be used not only to provide both an on-air picture and a preview picture in full graphic detail, but can lead the way toward future montaging of other effects on the graphics output channels.

This very elegant phrase is a reiteration of the determination that any equipment built by Chyron be upwardly mobile. As new requirements and new capabilities occur in the graphics field, we design our devices to be able to handle them.

For example, our new automatic display system allows the Chyron operator to precompose the sequence of messages and control the affects that are desired. By only a single button press, sequences are allowed to occur, either under a self-timed operation or on manual cue. This deals only with the character generator operation itself. Other devices being developed, which are deliverable this spring and summer, allow our character generators to be the controlling and integral heart of a complex graphics system.

These are the basic requirements. Because the field is new and the industry is just "getting its feet wet" in these digital concepts, we are beginning to establish standards which can be applied in measuring the utility of one graphics titling or character generator against another.

Because Chyron has responded to the needs of the industry, our program of continually upgrading and escalating the capabilities of our equipment has paid off in providing a leading group of components pointing the way towards even greater utility in the broadcast field.

Mr. Butler:

Thank you very much, Gene. Now Tom Meyer.

Mr. Meyer:

The function of the television industry is communication and entertainment. Television equipment exists only to facilitate that function and does not of itself communicate or entertain, disregarding, of course, the momentary wonder at large-scale integrated legerdemain or the fleeting excitement occasioned by self-immolation of a critical component in a forty plus share program.

The ENG revolution and each of the preceding broadcast revolutions have been provoked principally by the need to communicate more thoroughly and entertain more creatively. The character generator supports the process of communication and entertainment and is best used to enhance good programming.

As with other enhancers, however, the character generator can do little for an input signal containing limited information, and if applied too deligently to a good signal may well degrade the end product.

The graphics generator is, by function, a production tool. As with the ENG camera, it serves its owner best when used by production personnel. Indeed, the trend toward production operation of character generators is well established and will accelerate if production personnel strive to clarify information delivered, illustrate facts and figures with simple graphics, present large quantities of data associated with such events as sporting events, elections, and most importantly engrave a sponsor's identity and location in the mind of the viewer.

This trend imposes rigid technical constraints on the design and implementation of graphic systems. The operational techniques must be simple and quickly learned by nontechnical operators. The **Telemation Compositor 1** provides a highly developed interactive communication system to facilitate this need.

The equipment must be able to defend itself in potentially hostile environments. The end product must be equal in quality to the handiwork of a graphics artist.

The fast pace of today's programming, particularly the highly competitive local news show, demands the ability to continuously update graphics displays without disturbing operations in progress and the ability to preview the material in the exact format in which it is going to be displayed on air.

Precomposed graphic displays should be readily transferable from the studio to locations remotely located. Random access storage must be provided for large quantities of material for instant recall. The use of precomposed random scripts to effect single key recall is highly desirable.

The needs of the engineering department are equally demanding. The graphics system must fit into the existing video system with minimal effort. The technical skills available in the average station must be usable to maintain the system.

The system should accept several remote keyboards connected via small cables, preferably a single pair. It must contain on board facilities for rapid isolation of failures in the digital circuitry used by modern graphics generators. The components must be readily available from multiple sources and be of types which be expected to remain on the market for many years.

While the function of the television industry is entertainment and communication, the goal of the industry is and always must be profit. Equipment purchases must be justifiable by increased sales, decreased costs, or both.

The needs and uses for graphic systems are in a state of rapid and continuous change. The relatively large capital expenditure involved requires that systems be capable of adaptation to new techniques without modification of the base hardware or loss of revenue time. This capability is best provided by modern mini- or microcomputers used as integral components of the system.

System capabilities are best implemented in the computer program to allow rapid, logical growth. The basic graphic generator must be capable of graphic quality alphanumerics and camera grade graphics. Font and graphics composition must be possible within the base hardware system, preferably in simultaneous operation with alphanumeric display.

The ability to include tailored operating programs for specific production situations, such as sporting events and elections, is a must. Preferably, these tailored programs should be available for operator call-up within the system's storage.

Industry standard data interconnect facilities must be included in the base system to guarantee integration with editing systems, remote computers, modern video tape and disc control systems, data control switchers and new devices working just outside the compound.

At Telemation, we have exerted great effort to answer these production, technical and financial needs which have come from you, the user. Systems now available are capable of a high degree of program enhancement. As with all technologies, the needs of today, tomorrow and 1982 must be considered before embarking on the first leg of the journey.

Mr. Butler:

That concludes the presentations. Are there any questions?

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Question:

Tom, you spoke of a tailored program. Could you describe what one of those tailored programs might do?

Mr. Meyer:

An example of the tailored program would be our modified operating program which provides the 10-1 election system, which runs as an integral part of the Compositor 1.

This system first appeared five years ago. It was used in the presidential elections of that time and has been successfully used in elections since that point in time.

It now lives in the system and is indeed immediately recallable by operational personnel in lieu of the standard, everyday operating program.

•

Question:

Mr. Leonard, how would you compare the utility of a camera generated video signal against a digitally generated video signal?

Mr. Leonard:

I would think that there are probably about three different aspects of that question.

One, the nature of the information, or, rather, the content of the television channel. It comes in two forms. One is an emotional content. You have to be excited by a properly deodorized arm pit. And then comes the information content. You have to identify precisely which deodorant it is that's going to do the best job of deodorizing.

In the one case, there's no graphic capability yet, given the cost of memory. And that's the only criterion right now — the cost of memory.

There's no graphic capability yet which can do the arm pit job. There's no camera capability that can do the clean, precise display of complex logos that digital counted out video can produce. Their problem is that the graphics presently is limited to flat color, any color you like within that pattern; but it's limited to flat color.

The complexity of output that you can get with cameras involves a great deal more equipment, people and manipulation than you can get, for example, with dynamic graphics when a news flash comes in you can quickly get something up on the screen which will be attractive and point to your news program at 11:00.

•

Question:

It seemed to me that Mr. Hindle felt that the actual character generation isn't as important as the manipulative power, the ability to move anything on the screen. Is that true?

Mr. Hindle:

Once you've authentically copied an art card or a graphic or something, the next problem is how you get it up onto the screen. Let me go over to the audio world for the moment and make an analogy.

When digital or electronic technology offered the ability to recreate the sound of a piano or any other kind of musical instrument on a Moog synthesizer, the operator who could make that device fly, who could actually play the Moog synthesizer and make the orchestra sound was not a bassoonist, not a clarinetist but the piano player. Since the control board of the Moog synthesizer happened to require the skills of a piano player.

It's the same thing with a character generator. One of the reasons that, the Vidifont has been very successful is that the keyboard is relatively transparent to an operator. We're asking the operator to just be able to type and not make any subdecisions while he's getting that information up on the screen.

If you start adding additional complexities and you want him to do rolls, crawls, montaging and things like that, the operator's going to have to perform some other kinds of motions with his hands or her hands.

We've talked about interactive systems or prompting. The more prompting you need requires a certain level of quantifying, because it's no longer transparent. Even video tape or film editors have to make some kind of decision.

So it is with your character generator operator. The more he's called upon to make decisions, the less transparent the control.

We really have to know what skills are required in the television facility. It's a rare breed of person who can create graphics and type displays and still be creative in composition, but still be able to look and operate a keyboard.

Probably at most stations the graphic artist is not sitting down and doing all the nitty-gritty composition of a character generator. They're much more comfortable with a felt-tipped pen or a brush, or something like that, workong on paper.

•

Question:

Frank, what direction do you think character generator development is taking?

Mr. D'Ascenzo:

It's of prime importance to everyone in the room that the kind of equipment we develop is certainly usable for a long time within their studios. In order to make it that way, we're going to have to all work hard to make our systems simple.

We're into video topography and we almost have to separate the creative people from the operators. There are two different kinds of people. The fellow that could sit down and make the fingers fly over the keys — that's one sort of an individual. But you need a little bit of creative input.

That's where the systems are going. They're going towards devices and mechanisms to allow direct interactive communication between the creator and the devices themselves. We have to paint pictures: we have to make electronic palettes, etcetera.

I was talking with a young lady from one of the networks who is a creative topographer. She told us what we ought to be doing in order to make that system more usable by her. Those are the kind of inputs we certainly need. And I think we'll all agree that that's the thrust, the general thrust of the industry.

•

Question:

What relationship has been established between the technology and traditional animation industries, and how do they relate to the work that you've done?

Mr. Leonard:

First of all, we had to learn something about animation and, to do that, we had to work with people in the primarily educational television area. We had to know how much motion is required and how much time, and so on.

We have to learn what the Ox-berry machine did and why it is such a tedious thing to get an animation sequence up for a news broadcast using Ox-berry when you had a developed film. You couldn't possibly do it. We had to develop controls that allowed those kinds of selections to be made.

It brings up an area which one of our colleagues at KCMO has dubbed the "follow me" area. Follow me means that you spend all the time in the world, within the constraints of getting it on air in time, to fiddle around with the effects and then when they've got it right put it in digital sequence which will repeat identically with precision.

I hope later on somebody will touch on the problems of developing control codes, because we've got the same problem in switchers, in tape editing and in character generators. Things are too complicated for anybody to do it right the first time.

The advantage of electronics is nothing is permanent until you so indicate. We must not constrain the graphic artists. We've got to learn what they can do, find out what their measurements are, and then give them the ability to do it a lot easier.

Mr. D'Ascenzo:

We stumbled across animation and the requirements for it several years ago. It was in a television studio and a fellow was using one of our smaller machines, and doing some interesting things with it. He was making words run across the screen and up and down. And I looked at that, and said, "My gracious, with just a few little changes we could make that a standard feature." So we did.

The interesting thing is that there are two kinds of animation. There's a word animation and a picture animation. We have word animation currently available in our systems. And we can create character and words that move around, bounce up and down, etcetera.

Picture animation is something that's different again. I assume you probably were referring more to picture type animation perhaps than word type animation.

We had experimented about a year ago with a synthesizer type device and a sort of a frame store mechanism where we can point a camera at a black and white art card, store this in a memory, and then as you bring that up create a synthesized effect utilizing the white versus the black video levels and doing different things by video mixing technologies so that as you brought the picture up, it would seem to evolve into motion. This was sort of a pseudo animation. That's certainly a step we're thinking of taking.

Mr. Leonard:

There is no difference between a picture or a word. A graphic is a graphic. If you want to animate a shape and your machine is capable of fine enough movement or positioning and fine enough time control of the positioning, I couldn't care whether it's words, graphics, kangaroos or background: you can animate it. We do animate graphics including nonword pictures.

•

Question:

What are the problems between operators and creative talent?

Mr. Meyer:

We've had the interesting experience now at several stations, where the union permits, of having actual graphics people placed in control of the machine, along with several other types of people.

At one of our stations, in particular, an interesting division of labor has occurred. One of the many remote keyboards on the system is located in the graphics art department. A very good graphics artist prepares formats for many types of needs for the station. These are stored and protected in the system memory.

Our system is constructed so that once a format, font, color, background, arrangements have been placed in memory, a lower level technical operator may recall this page and type directly into the page without having to know how to perform graphics arts munipulations, formating, esthetics. New information is typed into the graphic, and it's stored elsewhere in the memory for use in that day's programming.

So there seems to be a trend to permit it to be a distinct division of labor that still very much involves the talents and services of the graphic artist.

Panelist:

I think we're in an area which is very specialized and many questions remain unanswered.

A division of labor is and does happen, and, will happen more logically in three areas: one being the guy who hits the switch and says "I have been instructed to cause program 1085 to occur at this instant." A second individual will be responsible for making sure that 1085 integrates the switcher character generator and any other equipment that was necessary into it. And thirdly, somebody who spent as much time as is available in creating the shape, the positioning of the various shapes, creating their colors and creating the sequence of motion they're going to go through.

Now one of the answers for the smaller station is to be able to obtain these shapes and sequences from their vendor as a service, so they can bypass the need for that highest level of creativity.

But I do think there's a lot of problems in character generators to be answered right on the first and second level of operation; namely the operator and how do you get in and out of the switcher. How do you get the right color to come through the switcher when you've got a character generator, and details of that sort.

•

Question:

What problems do you forsee in the interfacing of your special generation and graphic generation and the time sharing computer, both from the standpoint of transmission, and recall?

Mr. Meyer:

I think that's relatively easy. Just like a computer being entered into a television station for traffic control or accounting, the keyboard control is pretty much the same that those people are used to. And if they are used to entering prepared sequences such as titling or graphics during the creative process, putting it onto a computer for playback won't be that difficult. It's being done now, for that matter.

Mr. Leonard:

I want to disagree completely.

You said time share computer. You don't know when that computer is going to be loaded up and the thing you need back in a hurry is going to take 15 seconds to respond. I think there's a very serious problem.

We've done a lot of work with computers at remote distances, having to use telephone lines. Time sharing implies the disciplines of time sharing — and usually you've got to live with an IBM communication discipline. You've got to be careful that your vendor or somebody who's doing the job with you, understands the traffic, the cuing and how long it's going to take for the remote computer to respond to you in time to get it on your half-hour segment.

Mr. Meyer:

If you're requiring total random access back from time share, that will be a problem. If it is a sequence system with a buffer memory in your character generator once you get access you grab as much of that sequence as possible in a buffer memory within your station, that won't be as much of a problem. But total random access, yes, I'll agree.

Mr. Butler:

Obviously, we should have had three hours for this discussion. Thank you very much for the panel and myself for your attention.

The Application of Digital Techniques For Video Measurements

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Beaverton, Ore.

Unmanned automatic operation of television broadcast transmitters is about to become an operational reality in the United States. Automatic operation implies that automatic measurement techniques will be required in the future.

It is this problem which Tektronix, Inc., has responded to in developing ANSWER II. This is an automatic video signal parameter measuring system with logging capabilities.

Upon hearing the acronym ANSWER II, some of you will recall seeing a demonstration of its predecessor, ANSWER I, during the 1972 convention. That equipment was actually a remote vertical interval test signal (VITS) waveform monitoring and telemetry system. It sampled the VITS at 1024 points along the test line, taking one sample per frame. In 35 seconds, it scanned an entire VITS. The samples were telemetered to a storage oscilloscope, which we employed for remote display of the VITS waveform.

In our demonstration, station KOAP-TV in Portland, Ore., had the sampling equipment located in its master control, and we observed the VITS in Chicago (they were inserted at KCET in Los Angeles). It is interesting to recall that, at one demonstration, the terminating resistor was accidentally removed from our input. We saw double amplitude sync pulses in Chicago, telephoned immediately to Mr. Tony Schmidt, Chief Engineer at KOAP-TV in Portland, and asked him to check the termination. He was a bit confused because he thought we were in Chicago. We confirmed that we were indeed in Chicago at NAB, and he very quickly remedied the problem.

ANSWER I was not put into production, partially because it was premature in concept, and partially because its complex analog circuitry did not encourage its installation at unattended sites where no one could keep it properly calibrated.

We foresaw that when the NTSC video signal could be digitized, these measurements could be made digitally at far lower cost and with obviously improved reliability compared to when they were made by analog techniques. Accordingly, Tektronix concentrated effort on the development of an instrumentation-grade analog-to-digital converter (ADC) to interface the video signal to a digital computer.

During this time, the semiconductor industry developed very tiny microprocessor devices, which are really mini-computers on a chip. The advantages of these devices are in their size and greatly reduced cost. These technology advances led to the digital version of ANSWER I; ANSWER II.

How It Works . . .

ANSWER II is an automatic measuring system which can measure all parameters of transmission distortions of video test signals, video signal-to-noise ratio (S/N), and the timing of horizontal sync, blanking and burst.

These measurements can be made at whatever time intervals that may be selected by the user. Results can be logged automatically. The logging can take place at a site remote from the site at which the measurements are actually made. For example, the ANSWER II System may be installed either at the transmitter or at a place where clear line-of-sight reception is possible, and the data logging can be done in the master control room.

Generally, it is necessary to compare measured performance with a set of normal performance limits, and, whenever out-of-normal tolerance conditions occur, some sort of alarm may be desired. At such times, it is required to log the data, including time-of-day and average picture level (APL).

The customer can program the equipment in all of the following operational parameters:

a. Tolerances, plus or minus. Two sets of limits can be established, inner and outer.
b. Those parameters will initiate the alarm when they are out of tolerance.
c. Logging at either fixed time intervals, or only during out -of-tolerance conditions (or both).

Our Digitized System . . .

When we decided to digitize the video signal so that we could use digital signal processing to make these measurements and to deal with limit comparisons, etc., we had to decide on our digitizing scheme. This involved:

How many bits?

a. 8-bit resolution is 0.25%; considerably better than the resolution of conventional waveform monitors.
b. Use of signal averaging, combined with signal dither, extends our resolution by one additional bit.
c. We decided on 8-bits as being highly compatible with most available digital integrated circuits.

What would the sampling frequency be?

a. The choice was between $3 \times f_{sc}$ (color subcarrier), which was the prevailing practice at the time, and $4 \times f_{sc}$, which affords very significant performance advantages which apply especially to measurements.
b. We opted for the faster sampling rate, $4 \times f_{sc}$, even though this would require more memory and a faster ADC.

Where could we get a suitable ADC?

a. With the parameters defined, it was obvious that the ADC would have to be developed in-house. We knew it had to use integrated circuitry because of cost, power, size and reliability considerations. So we undertook our own in -house I. C. program, too. The ADC which resulted from this in-house development has been described by this author in a paper presented at the 1977 SMPTE Winter Conference.

One of the great attractions of digital signal processing is the ability to store large amounts of video information in a relatively cheap manner, without suffering distortions. This is useful in the present instance to reduce the masking effects of random noise, thereby improving accuracy.

Combined with signal dither, averaging gives us 9-bit resolution and up to 15 dB enhancement in video S/N as it relates to measuring purposes. No, we cannot use ANSWER II to reduce noise in the broadcast picture signal.

One further point about digital memories: Because their cost is rapidly declining, and we expect this to continue, we can afford to employ a great deal of memory in ANSWER II.

Digitized Measurements . . .

The truly all-pervasive justification for digitizing the signal to be measured is that a microprocessor can do the measurements. The measuring power of a microprocessor is limited only by the software provided. It replaces a mini-computer in this function, with an impressive cost savings and size reduction.

Sampling the $\frac{455}{2}$ cycles of color subcarrier per television line (NTSC) at $4 \times f_{sc}$ yields 910 samples per Line. Since commercial memory packages are convienently organized into 1024 bytes, they easily and economically accommodate storing a television line.

Memory capacity of ANSWER II will adequately store 32 NTSC television lines. These 32 television lines of memory capacity may all be used to store a chosen VIT Signal on 32 successive frames. This permits 32 stored samples, representing any point on that VIT Signal, to be averaged. Averaging over 32 samples has the effect of improving the accuracy of measurements, which is equivalent to an improvement in the S/N of ~15 dB.

FCC Rules for the visual transmitter specify 44 dB S/N (minimum). With 32-sample averaging, the apparent S/N is 70 dB. Rms noise voltage is 230 μV. Assuming a 15 dB quasi-peak-to-rms crest factor, the quasi-peak noise is -55 dB, or 1.4 mV. These noise peaks are exceeded only 1% of the time, and may be neglected. This noise level is insignificant, compared to the level of a least significant bit (LSB), 5 mV, at the input.

An alternate to using the entire memory for one VIT Signal is to allocate 16 lines of memory to each of two VIT Signals. This reduces the noise averaging by 3 dB without increasing the time required to make the measurement—still about one second. Further partitioning of the memory so that four VIT Signals can be averaged over 8 frames each results in noise averaging of 9 dB. The data is all taken in one second.

The VIR Signal can be monitored by the ANSWER II System by either memory partitioning or time-sharing first the VIT Signals for a second, and then the VIR Signal. The parameters which can be measured are listed in Figure 1.

For many applications, the most important transmission parameter is video S/N. This is accomplished by measuring the noise power on an unused line in the vertical blanking interval. (FCC Rules prohibit use of lines previous to line 17, so far as the radiated signal is concerned.)

ANSWER II can be programmed to measure the noise power on any of these unused lines. The microprocessor computes the rms value of the noise occurring within 52 s on the chosen line, and then it calculates the S/N using the bar amplitude as the reference amplitude. The user selects the noise test line via his keyboard.

Noise is measured over a specific bandpass, normally 4.2 MHz. In analog measurement equipment, a low-pass filter is used to restrict the noise bandpass. In a digital measurement system, the bandpass may be restricted by use of digital filtering techniques. Digital filtering is a matter of software programming.

For measurements of long distance radio relay systems (both terrestrial and satellite), noise is measured with a weighting filter and a 5 MHz low-pass filter (Draft Rec. AB/CMTT[1] and EIA RS250-B). The loss versus frequency-gain characteristics can be realized using digital filtering techniques, i.e., proper software. The implications of digital filtering are:

- Analog filters are not required.
- No question arises as to their alignment, either in manufacture or in the field.
- Signal re-routing through filters to measure noise is not required (such analog circuitry might introduce distortions).
- Should different filtering requirements ever arise, the software is changed, which amounts to replacement of the relevant Programmable Read-Only Memory (PROM) integrated circuits.

TRANSMISSION PARAMETERS

FOR AUTOMATIC MEASUREMENTS

Insertion Gain

Video Signal-to-Noise Ratio
(C.C.I.R. Unified Weighting to 5 MHz)

Sync Amplitude

Burst Amplitude

Line-time Linear Distortion

Short-time Linear Distortion

Bar Trail (625/50 Standards only)

Pulse-to-Bar Ratio

2T Pulse, K or P factor

Luminance Non-linear Distortion

Differential Gain

Differential Phase

Burst Processing Phase Error

Chrominance/Luminance Gain Inequality

Chrominance/Luminance Delay Inequality

Chrominance/Luminance Intermodulation

Figure 1.

Digitized Monitoring . . .

A broadcaster may wish to use one ANSWER II System to monitor his transmitter's signal off-air, and to monitor the incoming feed from the network.

There are differences in detail between the VIT Signals which are required to be radiated by broadcasters operating under remote control authorization, and the national VIT Signals inserted by all networks. For example, the composite signals differ in order of presentation of staircase and line bar elements, and the FCC signal has 2T integrated, sine-squared transitions versus 1T for the network signal. The network multiburst occupies only a portion of the test line, followed by the "pink panther" three chroma level signal. These differences are handled by software, i.e., additional PROM's so that when ANSWER II is instructed to monitor incoming or radiated VIT Signals, the appropriate PROM's control the microprocessor. The additional PROM's required represent only a small fraction of the total used.

In addition to monitoring VIT and VIR Signals, ANSWER II is capable of monitoring the timing relationships in the blanking interval.

H SYNC, H BLANKING, BURST TIMING PARAMETERS[1]

Timing Parameter	From	To
Front Porch[2]	+ 4 IRE	− 20 IRE, leading edge sync
Sync Width[2]	−20 IRE	− 20 IRE, across sync pulse
Sync to Blanking End[2]	−30 IRE	+ 4 IRE
Burst Width[2]	0 IRE	(following that ≧50% burst)
	0 IRE	(following that last peak ≧50% burst amplitude)
Breezeway[2]	−20 IRE	0 IRE burst start point
Sync to Burst End[2]	−20 IRE	0 IRE
Color Backporch[2]	0 IRE	+ 4 IRE
Blanking Width[2]	+ 4 IRE	+ 4 IRE

[1]In accordance with EIA Recommended Practice for Horizontal Sync, Blanking and Burst Timing in Television Broadcasting.
[2]Measured in nanoseconds.

Figure 2.

Figure 2 shows the timing parameters which can be measured in accordance with EIA Recommended Practice. These parameters are read out in nanoseconds at the half amplitude points as recommended by The Broadcast Television Systems Committee.

Resolution of these measurements is not limited to +70 ns, as one might infer, but extends, through digital interpolation, to a few nanoseconds.

One of the most difficult measurements to make is SC-H timing. While present FCC Rules define this time relationship as random, there is an urgent need to define this relationship in any signal that is to be video tape recorded.

The Broadcast Television Systems Committee has proposed definitions for the four distinct color fields which exist in NTSC signals because of the half-line offset between luminance and chrominance spectral components.

Figure 3 shows color fields I and III. The difference between them lies in the sense of the subcarrier at the zero crossing of the color sync burst which follows a peak ≧50% burst amplitude. A signal is said to be SC-H timed if the time between the 50% point on the leading edge of H sync and this zero crossing is equal to the period of 19 ± 1/8 cycles of subcarrier.

Nineteen cycles at 3.58 MHz = 5.307 s, but this must be measured to less than ± 0.6% or ± 35 ns, which is not possible using the best television waveform monitors. The problem is considerably simplified in ANSWER II because the incoming signal is sampled at $4 \times f_{sc}$, and half of these samples occur precisely at zero crossings of color sync burst.

ANSWER II, having evaluated the sample immediately preceding that zero crossing which exceeds half of burst amplitude, looks into the memory, 4 × 19 = 76 samples earlier, and evaluates sync at that instant. If the sync amplitude at that instant equals one-half sync amplitude, we have zero error in SC-H timing.

In practice, linear interpolation between sync amplitudes one sample earlier and one sample later enables us to measure actual error in SC-H timing. By determining whether the sample of burst taken just before the zero crossing is positive or negative, we identify each of the four fields.

If we derive our sampling pulses from color sync burst on the broadcaster's black burst signal, we can determine whether any other video signal presented to the ADC of ANSWER II has the same (or the opposite) color field as plant sync, and, in all cases, the SC-H timing error can be read out.

This may appear to be an elaborate means to measure SC-H timing, yet all it requires is the correct PROM's within ANSWER II that will instruct the microprocessor what to do. It is also an example of a measurement capability which is not readily accomplished using conventional analog techniques, and it does not require an engineer to make the measurement.

A Management Tool . . .

Now that we have discussed what can be measured, it is appropriate to consider the use of the measured data by the broadcaster.

First, a broadcaster needs to know that his radiated signal conforms to FCC Rules. Second, he wants to know the quality of his signal, hence the quality of the picture available to his audience. Third, he wants to detect changes in the performance of his transmitter.

There is, then, a need to compare the measured values of certain parameters with limits related to those prescribed in the rules. With ANSWER II, the microprocessor carries out such comparisons for the broadcaster, and indicates out-of-tolerance conditions. The broadcaster programs his system as to the limits he wishes to establish, and communicates these limits to ANSWER II via a keyboard.

When an out-of-tolerance situation occurs, it is automatically logged, together with the time of day and the average picture level (APL). Because it is quite normal for nonlinear distortions to increase at extreme APL conditions, and especially immediately following large changes in APL, the system does not initiate an alarm unless the out-of-tolerance condition is found to persist.

ANSWER II will be the first commercially available digital computer system designed to monitor broadcasters' test and reference signal automatically. It will relieve engineers of much of the tedious and repetitious work so that they can devote their talents to more productive efforts. It is a management tool because it can log all these measurements so that the broadcaster can detect changes in the operation of his transmitter.

Eventually, as broadcasters gain familiarity with and confidence in automated measurements, many other uses will be found for this system.

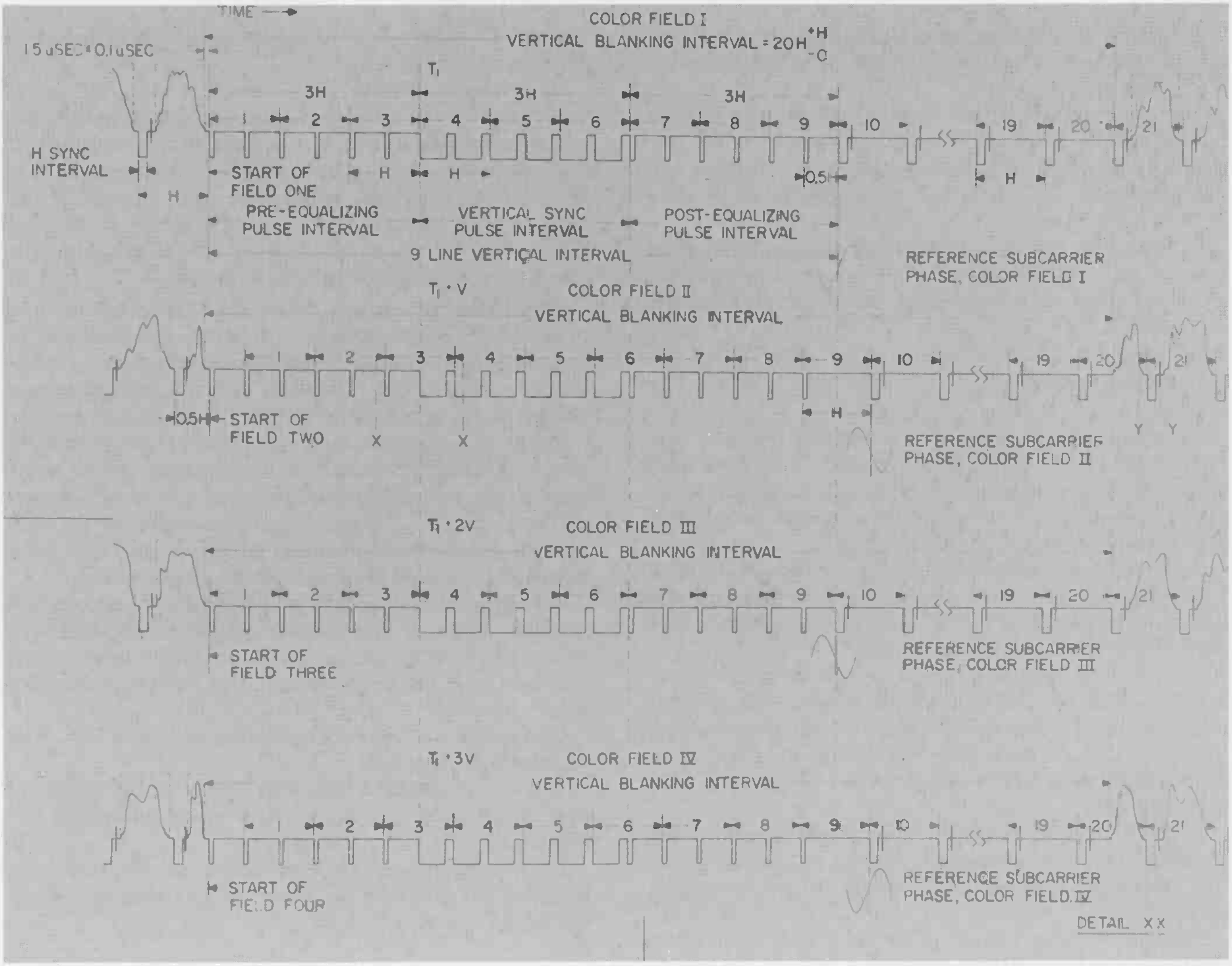
TIME
COLOR FIELD I
VERTICAL BLANKING INTERVAL = 20H +H -0
1.5 uSEC ± 0.1 uSEC
3H
3H
3H
H SYNC INTERVAL
START OF FIELD ONE
PRE-EQUALIZING PULSE INTERVAL
VERTICAL SYNC PULSE INTERVAL
POST-EQUALIZING PULSE INTERVAL
9 LINE VERTICAL INTERVAL
REFERENCE SUBCARRIER PHASE, COLOR FIELD I
COLOR FIELD II
VERTICAL BLANKING INTERVAL
START OF FIELD TWO
REFERENCE SUBCARRIER PHASE, COLOR FIELD II
COLOR FIELD III
VERTICAL BLANKING INTERVAL
START OF FIELD THREE
REFERENCE SUBCARRIER PHASE, COLOR FIELD III
COLOR FIELD IV
VERTICAL BLANKING INTERVAL
START OF FIELD FOUR
REFERENCE SUBCARRIER PHASE, COLOR FIELD IV
DETAIL XX

Figure 3.

A Report on TV and FM Rebroadcast Translators After 20 Years

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It is the intent of this paper to present a status report on rebroadcast translators 20 years after the first FCC authorized installations.

In this time we have seen UHF TV translators grow in power from the original 10 watts to 100 watts as the predominant size, VHF translators achieve respectibility and FM translators join the family on roughly the same terms as VHF translators.

Color transmissions, a curiosity to most in 1957, must now be faithfully handled by translators—in many instances through four repetitions, and in a few instances through five.

In numbers in the United States alone there are approximately 2300 VHF, 1100 UHF and 250 FM translators.

Rebroadcast translators were originally and still are, for the most part, heterodyne repeaters—i.e., frequency change by a heterodyne process and rebroadcast at an appropriate output power. However, in some exceptional situations a modulator is included.

Scattered but persistent grass roots activity in the mid-1950's led to the formal beginning of recognized and licensed translators in 1957. These were UHF units with outputs confined to channels 70 to 83.

The ensuing three years saw a modest installation rate of UHF translators, along with a stubborn disagreement between the FCC and the operators of many hundreds of VHF translators who were unwilling to forego the simplicity of their VHF equipment just because the FCC rules said UHF.

In 1960 the FCC rules were amended to permit VHF translators with one watt power output. In the mid-1960's this limit was increased to ten watts, where it remains today.[1]

In 1970 FM was added under rules which are quite similar to those for VHF translators.

Widespread Use . . .

Today, translators are operated by broadcast licensees, such local governmental units as cities, counties or special improvement districts, and service clubs, land developers and individuals.

The heaviest concentration is in the West but there are significant numbers in the Appalachian Mountains and along the southern tier of New York, in upper Michigan, Wisconsin and Minnesota.

As shown in **Figure 1** broadcasters operating translators are, for the most part, concerned with getting the signal from their station into a particular community or shadowed area. With a few exceptions, they concentrate on translators rebroadcasting their own stations. A few small market stations have a broader policy of helping local groups with their translator problems, including the translators rebroadcasting other primaries.

Originally, translators other than those owned and operated by licensees were purchased and installed by nonprofit local groups. Some groups were already established, such as local Chambers of Commerce or service clubs. Others were formed specifically for the purpose.

Local organizations conducted fund-raising drives to purchase, install and maintain equipment. While many such groups are still in operation and some new ones are still appearing, it is becoming more common to see local government funding providing television, and to a lesser extent FM, to white areas as a government function.

When the taxing authority of a local government unit is utilized to assure that all *users* support the cost of installation and operation the tax per family becomes very small, in most instances between five and ten dollars per family per year.

The power levels presently allowed under the FCC rules are shown in **Figure 2**.

The circumstances under which a TV or FM translator may be installed by a broadcast licensee is illustrated in **Figure 3**.

CATV operators are permitted to be translator licensees but generally not in the same community where the CATV system is located.

A Variety of Uses . . .

Multiple-hop translator systems are common and in some instances the vast majority of the population served is watching the last translator in the chain. The intermediate translator(s) must serve some population, but it is not uncommon for this to be a thinly scattered population in a rural area.

Smaller market stations have found it advantageous to install translators to bring outlying communities into their audience count. A typical example is shown in **Figure 4**.

1. VHF translators operating on an assigned channel can use 100 watts.

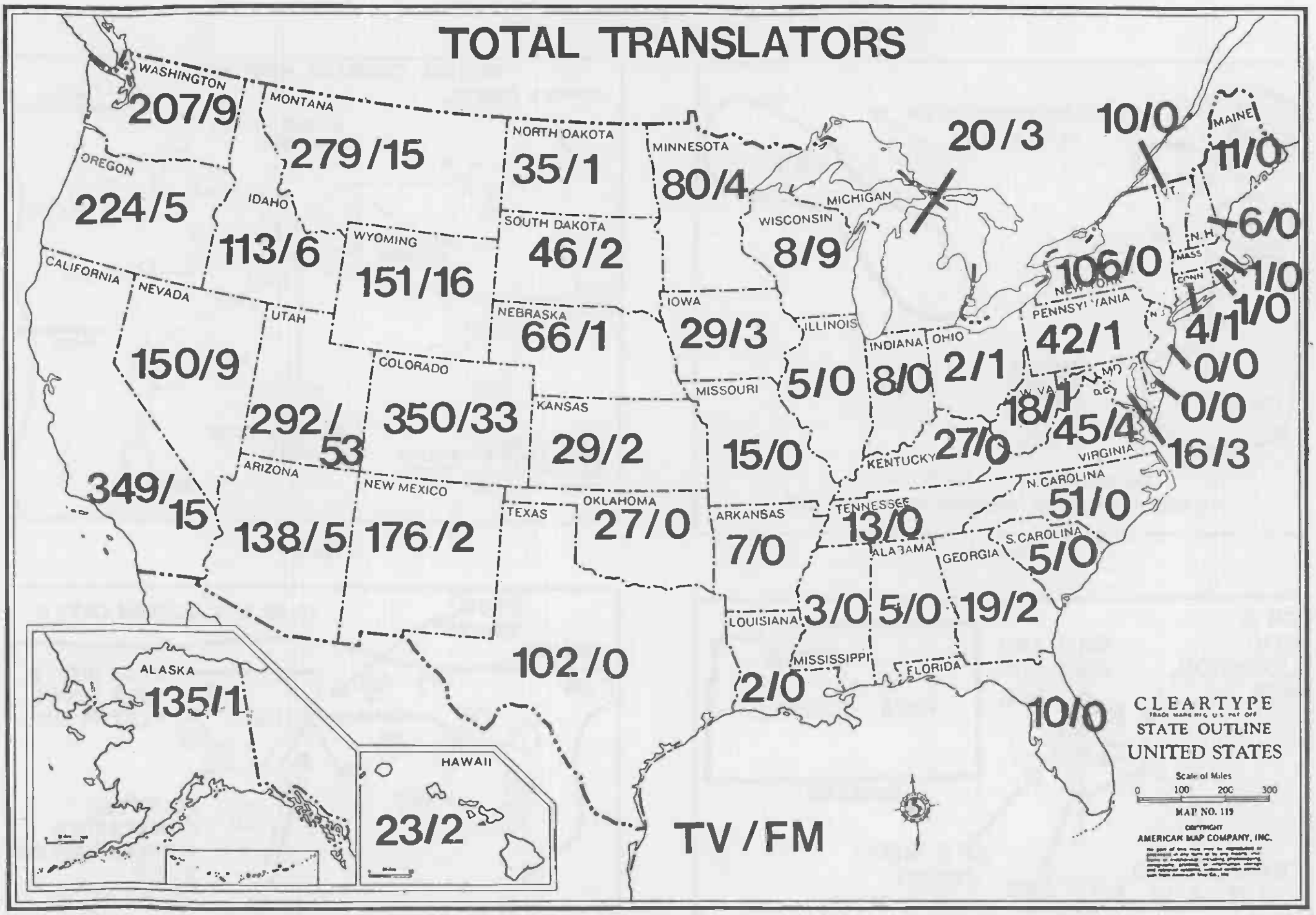


Figure 1.

POWER LEVELS – TRANSLATORS IN U.S.A.		
	SERVE AREA OR COMMUNITY EAST OF MISSISSIPPI	SERVE AREA OR COMMUNITY WEST OF MISSISSIPPI
VHF OUTPUT	1.0 WATT (ON CHANNEL LISTED IN TELEVISION TABLE OF ASSIGNMENTS – 100 WATTS)	10.0 WATTS
UHF OUTPUT	100 WATTS (ON CHANNEL LISTED IN TELEVISION TABLE OF ASSIGNMENTS – 1000 WATTS)	100 WATTS
FM	1.0 WATT (1 WATT LIMITATION ALSO APPLIES TO ZONE I-A)	10.0 WATTS

Figure 2.

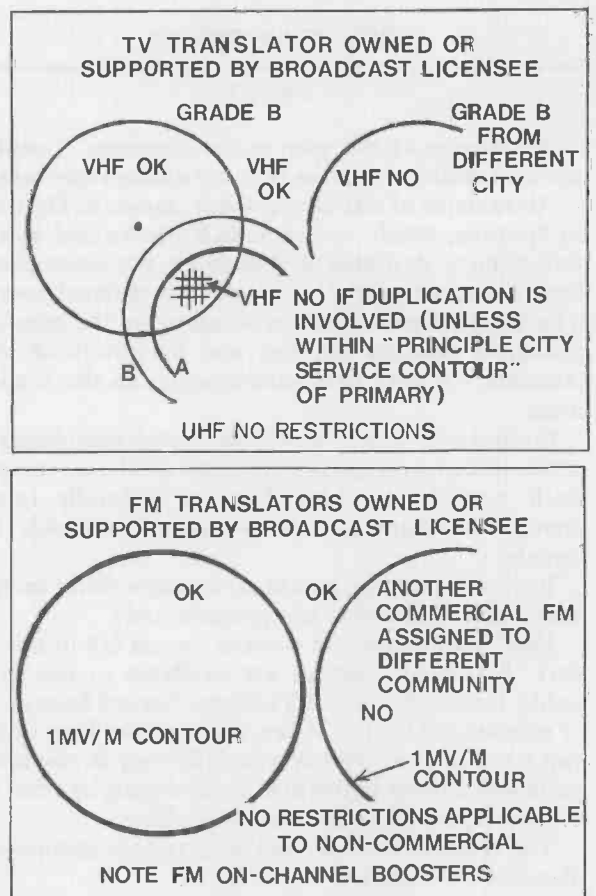


Figure 3.

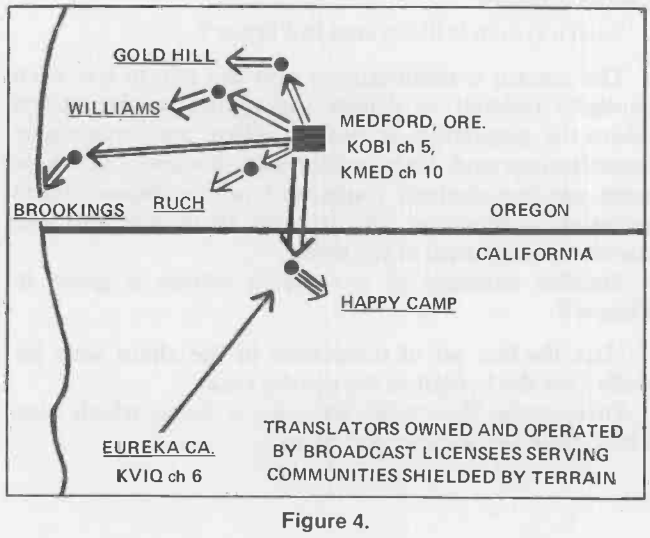


Figure 4.

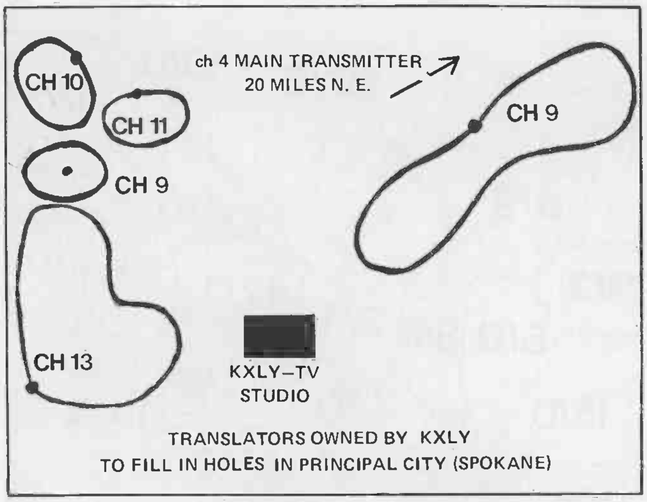


Figure 5.

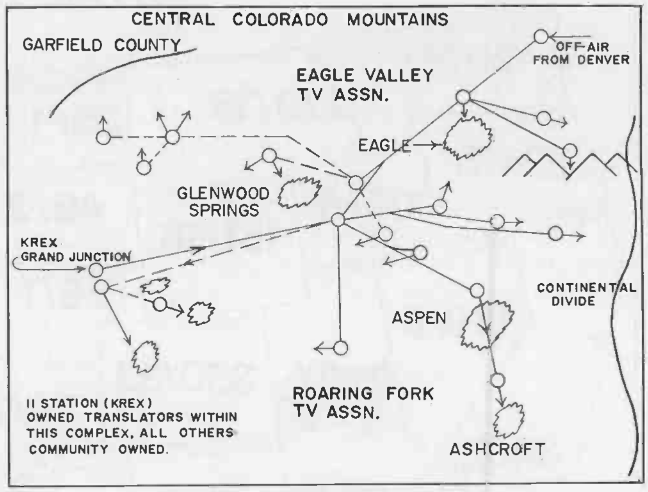


Figure 7.

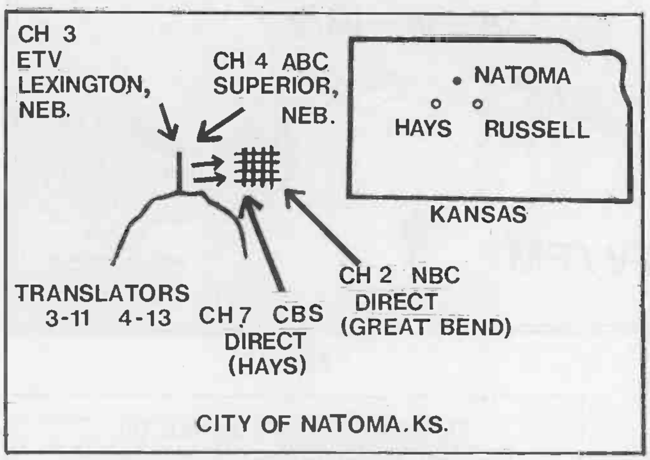


Figure 6.

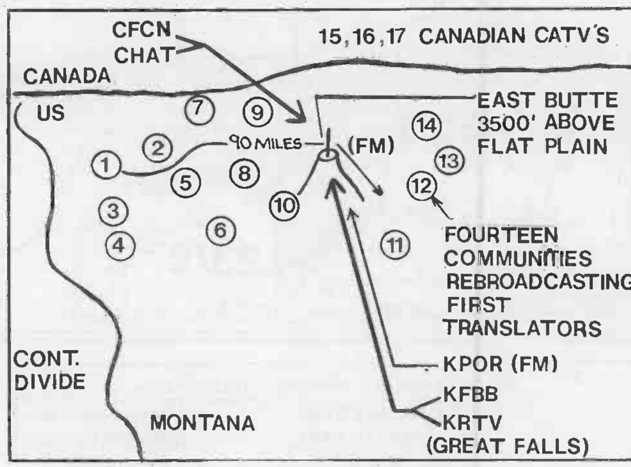


Figure 8.

A variation of this plan is also common. Translators are used to fill holes in the primary station's coverage.

An example of this arrangement, shown in **Figure 5**, is in Spokane, Wash., where KXLY has located its transmitter on a desirable well-elevated site some distance from the center of the city to get full regional coverage. The regional coverage was obtained at the expense of shadowed areas in the city, and a network of several translators is used to obtain coverage in the shadowed areas.

Community-owned translators came into being first where there was no consistent direct off the air reception. Such communities have by now generally installed enough translators to rebroadcast all available input signals.

Interest is building now in areas where direct reception is available from one or two primaries only.

Many communities in western Kansas fall in this category. Satisfactory signals are available to the general public from NBC and CBS stations, but not from an ABC or educational station. A few communities have installed two translators to remedy this deficiency in the last two years and a trend in this area is developing as other communities learn they can solve this problem.

The channel arrangement in a typical community is illustrated in **Figure 6.**

More dramatic are some of the large multi-hop TV systems. The one which has received the most attention is the Utah backbone system which, with its branches, makes the four Salt Lake City channels available to nearly every community in the southern part of Utah. The residents of this area have been fortunate to receive considerable support and assistance from the primary stations.

In some other areas complex systems have evolved without any support or encouragement from the primaries involved.

Such a system is illustrated in **Figure 7.**

The terrain is mountainous and the system has been carefully tailored to deliver the signals to the valleys where the population is located. There are three major organizations and many minor ones involved. Of these some use tax derived funds and others depend upon voluntary collections. Incidentally there are also FM translators at several of the sites.

Another example of a complex system is given in **Figure 8.**

Here the first set of translators in the chain were installed for the benefit of the nearby area.

Fortunately they were set atop a butte which rises about 3000 feet above a flat plain.

Communities on the plain and in the hills around the rim have discovered that their best input signals are from the first set of translators. Most existing translator stations have switched over to this input and new ones have been installed taking advantage of this signal source.

It might be noted in passing that a significant number of translators in the northern part of the United States utilize inputs from Canadian stations, even though the Canadian government prohibits the reverse of this.

Some Trend Setters . . .

There are some systems using techniques not yet common but which are trend setters and therefore worthy of note.

The first of these is solar-powered.

In the past, translator sites have represented a compromise of three factors; quality of input signal, ability to transmit to the area of interest, and power to operate the equipment. While a few translators have been installed using thermoelectric generators or engine driven generators, most have been confined to locations where power lines existed or could be built. In the latter case, this often was done at very large expense.

Now that solar power has become practical, one of the constraints has been removed and we have supplied complete solar powered packages for locations in Alaska, western Canada, many western states and northern Mexico.

A typical example is shown in **Figure 9**.

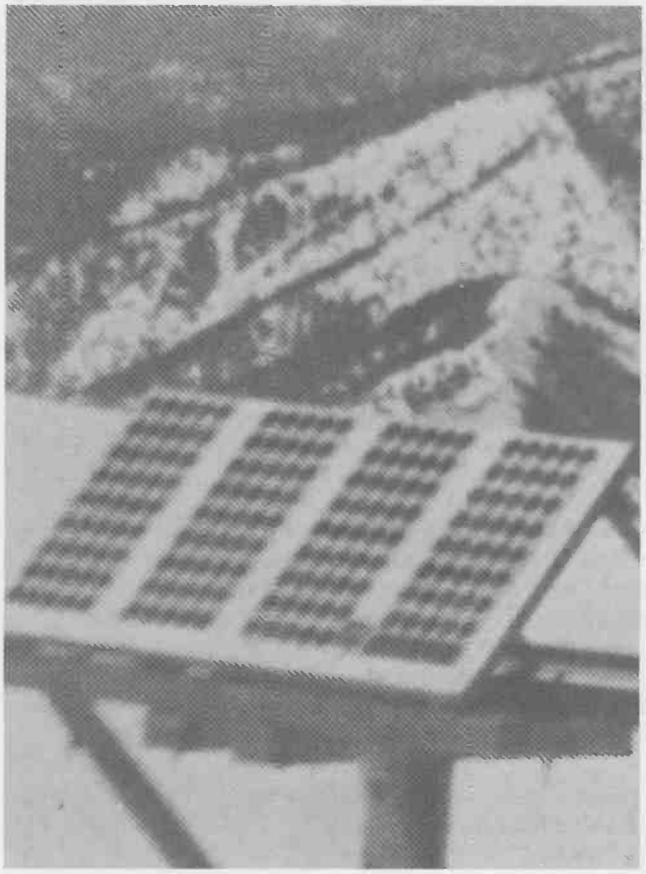


Figure 9.

The economic feasibility of the solar powered translator depends upon an efficient ratio of operating power to output signal as well as upon the improved cost per watt of solar panels which we have seen in the last several years.

The power flow is shown in **Figure 10**.

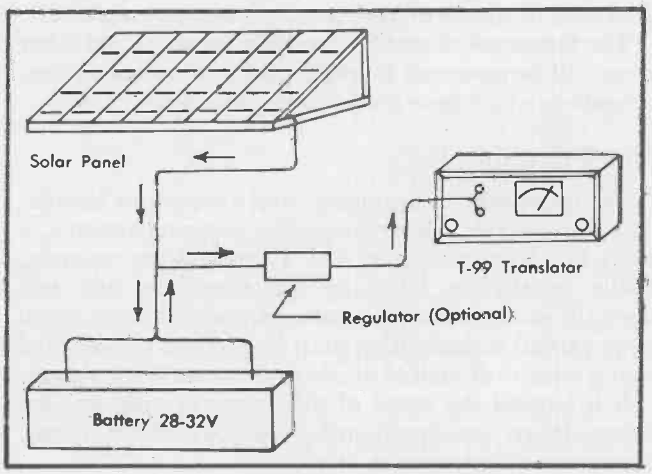


Figure 10.

Part of the success in this type of installation comes from a VHF translator design which produces a one watt peak sync output with a 3.5 watt d.c. input and a companion low drain 5 watt amplifier. The brochure from which the Figure 10 illustration was taken outlines the detailed engineering considerations necessary for use of solar power. The brochure is available on request from Television Technology Corporation.

The second type of translator installation in this trend-setter group uses microwave to deliver the signal to the translator site.

FCC rules for many years have contained provisions for AM heterodyne microwave in the 2GC band to be used for this purpose. In practice this has never proven to be a useful technique, suffering from the same limitations as multiple-hop translator systems and being more expensive.

To overcome the technical limitations of AM heterodyne repeater systems and to make use of the highly developed techniques and hardware which are available for conventional FM microwave installations, the use of this latter type of equipment has been requested on a waiver basis on several occasions. A proposed rule making to permit this type of operation is in process at the FCC and reportedly is near a final decision.

Some examples of multiple-hop translator systems have been presented earlier. There is a need to provide better and more consistent quality at the far end of such systems. This is the basic premise of the proposed rule making and it appears that there is little likelihood that an improved transmission medium would result in translator systems extending to greater distances.

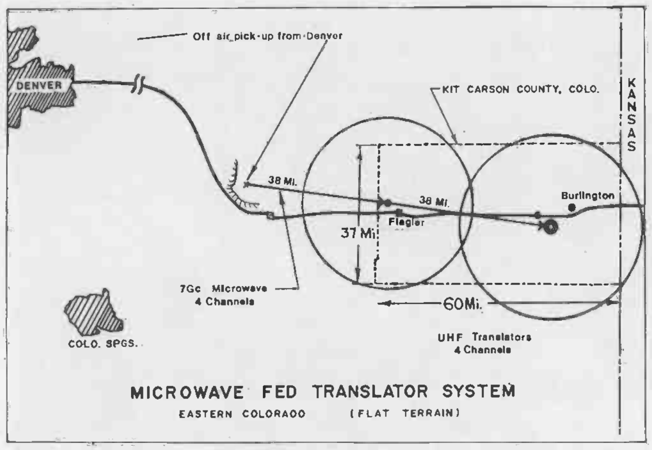


Figure 11.

An example of microwave-fed system which has recently been constructed is given in **Figures 11 and 12.** Note that a modulator is required as an interface between the output of the microwave receiver and the input of the translator.

In Alaska there are about 100 mini-stations where a transmission medium is tape and airplanes. Programs are rebroadcasted in isolated areas with a one-day delay.

Looking further in the future there is the possibility of bringing input signals to translators by the use of satellites. Technically this is little different from using FM microwave.

Two translators were included in the demonstration conducted by the Federation of Rocky Mountain States and HEW using the ATS-F satellite. And plans are underway in Alaska to feed its mini-stations by satellite.

The future use of satellite feeds for translators in other areas will be governed by regulatory and economic considerations which have not yet come into focus.

Not That Simple . . .

At first glance, it may appear that a translator installation is an uncritical assemblage of a receiving antenna, a black box (the translator) and a transmitting antenna. While installation following this simplistic view will normally produce limited results, in most instances much more careful consideration must be given to the detailed characteristics of each of the three elements of the system.

It is beyond the scope of this paper to give detailed system design considerations for each part of the system, but a few highlights are in order.

The reception problem consists of developing an interference-free, ghost-free, reasonably steady input signal without co-channeling.

In the case of translators, this is complicated by the presence of the output signals 80 to 120 db higher than desired. Frequently there are several translator outputs at the same location and it is not uncommon to find land-mobile base transmitters there, too. These latter transmitters cause interference which is difficult to diagnose because of their intermittent and irregular duty cycle.

In flat terrain where a significant tower is a necessity, the transmitting and receiving antennas are almost always on the same tower with little isolation between.

FREQUENCY PLAN
KIT CARSON COUNTY
MICROWAVE-FED
TRANSLATOR SYSTEM

ORIGINATING STATION	MICROWAVE CHANNEL	FLAGLER TRANSLATOR OUTPUT	BETHUNE-BURLINGTON TRANSLATOR OUTPUT
CH 4	6950-6975	63	53
CH 6	7000-7025	65	55
CH 7	7050-7075	67	57
CH 9	7100-7125	69	59

Figure 12.

FM translators are being added at existing TV translator sites with some regularity.

The second harmonics of the frequencies in the FM band fall neatly in channels 7 - 13, and FM translator output frequencies must be chosen with an eye to the potential conflict with the reception of high band TV channels.

To conquer this large collection of potential interferences, it is necessary to use, to the extent possible, terrain isolation, separation, directive antennas cancellation through phased antennas, preamplifiers and filtering.

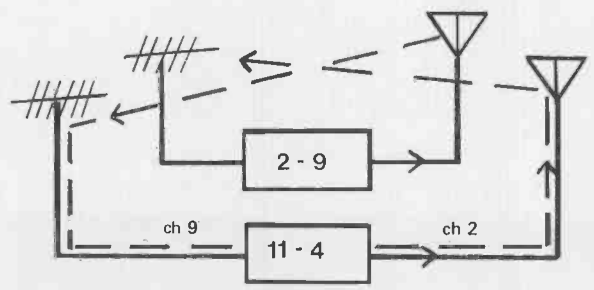
Occasionally the incoming signal will be unusable at the optimum re-transmitting point and it will be necessary to transport one or more signals a considerable distance. This occurs most commonly with a flat top mountain, where examples may be found with a mile or more of cable between the headend and the transmitting point.

It falls to the translator itself to process the signal. It must provide most, if not all, of the system selectively. AGC is its responsibility. It also must make the channel conversion correct the visual-aural ratio, if necessary, and develop the required output power without exceeding an acceptable limit on spurious radiation.

The spurious output problem is only infrequently a matter of concern in originating transmitters. By contract, economic considerations pertaining to translators dictate a common output stage to handle both the visual and aural signals. As a consequence inter-carrier beats are generated, which persist to some degree through the output filtering.

A decreasing but identifiable spectrum of spurious products can be found at multiples of 4.5 MHz below the visual and above the aural carrier.

If it is necessary to receive the second adjacent channel to the output at the same site, special precautions are in order.



EXAMPLE OF MUTUAL INTERFERENCE INVOLVING TWO TRANSLATORS

ANY ch 9 SIGNAL WHICH FORCES ITS WAY THROUGH THE 11-4 TRANSLATOR WILL COME OUT AS ch.2

Figure 13.

If it is necessary to receive a channel which is immediately adjacent to an output, very special precautions are in order. There are a number of installations where this is done.

It also is necessary to be on the alert for problems arising from the presence of the strong output carriers at the input of the same or a co-located translator. The problem may arise from brute force overload of the input stages of the affected translator or in some more subtle way.

An example of this is illustrated in Figure 13.

The transmitting antenna, single or array, is called upon to deliver the signal to the desired coverage area with maximum effectiveness.

Compared to the power levels of normal transmitters, translators operate at very low power indeed. Their success depends heavily upon using the available output power carefully and this, in turn, depends upon selecting or synthesizing antenna arrays with patterns which will concentrate the available energy where it is needed.

Over the years, a substantial collection of complete antennas and modules which can be used to assemble quite predictable arrays have become available.

The common types are summarized in Figure 14.

Stations with religious formats similarly command very high audience loyalty and have been relatively successful in finding local groups which are interested in becoming the licensee of a translator in their community.

There are a modest number of FM translators licensed to the primary station. These are used to fill in gaps in some instances and to extend coverage to white areas in others.

Overall the number of places where FM translators are in use is small compared to the number of places where they could solve problems and/or be profitable to the primary station. It appears likely that a considerable expansion in the number of operating FM translators should take place as more broadcasters and local groups discover their usefulness.

COMMON TRANSMITTING ANTENNAS VHF–TV

YAGI :	(5 TO 12 ELEMENT) GAIN ABOUT 10 dB OVER DIPOLE. HORIZONTAL COVERAGE 40°–70°. MAY BE VERTICALLY STACKED FOR HIGHER GAIN. TWO OR MORE MAY BE SKEWED IN AZIMUTH FOR WIDER HORIZONTAL ANGLE.
LOG PERIODIC :	SIMILAR TO YAGI, BUT SLIGHTLY LESS GAIN. BETTER IMPEDANCE MATCH AND ANTENNA CHARACTERISTICS ARE LESS UPSET BY ICE ACCUMULATIONS. SEVERAL CHANNELS MAY BE COMBINED.

COMMON TRANSMITTING ANTENNAS UHF–TV

CYLINDRICAL REFLECTOR & PARABOLIC	GAIN ABOUT 17 dB TO 24 dB (DIPOLE) HORIZONTAL COVERAGE 30°–5° DEPENDING UPON SIZE.
PANEL ANTENNAS	MODULAR, CAN BE USED TO SET UP ARRAY TAILORED TO COVERAGE REQUIREMENTS. INDIVIDUAL MODULE 14 DIPOLES IN FRONT OF BACK PLATE 9.5 dB WITH 90° HORIZONTAL COVERAGE.
SLOT ANTENNAS	NORMALLY AVAILABLE AS 8-BAY WITH GAIN OF 11½ dB OR 16-BAY WITH 14 dB GAIN. BASICALLY AN OMNIDIRECTIONAL ANTENNA BUT PATTERN SOMETIMES SHAPED TO ACHIEVE DIRECTIONAL CHARACTERISTICS. LENGTHS TYPICALLY 15 FT. FOR 8-BAY. MUST BE PRECISELY VERTICAL DUE TO NARROW VERTICAL BEAM WIDTH. DOWN TILT OF 1–2° COMMON. NULL FILL READILY AVAILABLE.

MULTIPLE CHANNEL OPERATION INTO ONE UHF TRANSMITTING ANTENNA COMMON. BANDWIDTH TYPICALLY ADEQUATE FOR 3 ALTERNATELY SPACED CHANNELS, eg. 60 62 & 64

FM TRANSLATOR TRANSMITTING ANTENNAS

YAGI :	SAME AS VHF–TV.
CROSSED DIPOLES :	OMNIDIRECTIONAL, MAY BE VERTICALLY STACKED, GAIN OF +3 dB (DIPOLE) WITH FOUR.
VERTICAL DIPOLE :	VERTICAL POLARIZATION, MAY BE STACKED FOR ADDITIONAL GAIN.
SLEEVE TYPE :	VERTICAL POLARIZATION, OMNIDIRECTIONAL.

Figure 14.

It is common practice to combine two, three or even four channels into a single transmitting antenna. Filters to do this are readily available as standard products for both VHF and UHF.

The number of channels which can be combined is usually limited by the bandwidth of the antenna.

FM translators appear to require omnidirectional transmitting antennas in a higher percentage of installations than VHF TV.

When horizontal polarization is required, good results have been obtained with crossed dipoles. A four bay array like the one illustrated in **Figure 15** can be assembled on a 20-foot mast, which is an economical size to work with.

My company has made one installation where omnidirectional coverage and vertical polarization were required. A four bay sleeve type antenna was used. The antenna was scaled up from a standard design the manufacturer had for a 150 MHz communications antenna.

There has been some casual interest in dually polarized FM antennas but so far no solution has been found which is compatible with the budget for an FM translator installation which is almost always under $5,000.

Where FM Stands . . .

It appears that FM translators so far have gotten off to a slow start.

A few Western stations thoroughly familiar with translators from their TV activities seized upon FM translators and made or encouraged their installation at existing sites. So far, much of the rest of the interest has centered around stations with specialized programming.

Stations with a classical music format seem to have the kind of determined audience which will, in an area of marginal reception, form a group to own and operate a translator. For instance, KVOD in Denver is currently carried on four FM translators, only one of which is station-owned. Classical music enthusiasts in two additional communities are actively talking about raising money to make similar installations.

Trends in TV . . .

In the TV arena we are seeing a continuing demand from rural residents for more channels and better quality.

There is a steady trend away from volunteer organizations operating on donations to some form of tax support which is making it financially practical to meet the demands.

Figure 15.

Within the last year, for the first time, we have begun to see two, three and even four counties meeting to discuss cooperative action on a regional basis.

I am hoping that it soon will be possible to offer such regional organizations an FM microwave backbone system so that near-in quality can be maintained to the far end of the system. Such an organization will be able to provide a budget which is up to the job.

There has been much discussion in recent years, particularly in government policymaking circles, about the problem of bringing a full complement of television to rural areas. The rural areas are truly beginning to solve the problem themselves and it appears the trend will accelerate.

In the 20 years since the official beginning of translators the techniques has progressed from a near curiosity to being a major element in the nation's TV and FM distribution.

New Concepts In All Solid State AM Broadcast Transmitters

Leonard L. Oursler
and
David A. Sauer
Broadcast Transmitter Engineering
RCA
Meadow Lands, Pa.

The history of amplitude modulated broadcast transmitters with electronic amplification devices began almost immediately following the invention of the triode vacuum tube around 1906.

AM broadcast stations progressed from operating powers of a few watts to 50,000 watts and higher within a few years as higher power tubes were developed.

As the electron tube transmitters grew in power output levels, it was not uncommon to achieve higher power by paralleling several power tubes. One example of this was an early RCA one kilowatt transmitter which utilized four tubes, each rated at two hundred and fifty watts, in a push-pull parallel circuit to achieve the one kilowatt of RF output.

Today, a single tube is capable of producing several million watts of radio frequency output.

In the modern world of solid state electronics, transistors have replaced vacuum tubes in more and more applications.

The AM broadcast transmitter is now entering the domain of solid state engineering. The design technology necessary for producing an all solid state broadcast transmitter has been available since the early 60's, but it was not until recently that the required higher powered transistors became available. It is now possible to produce large amounts of RF power by combining these solid state devices into transistor arrays.

Solid State Versus Tube Transmitters . . .

Much can be said about the differences between all solid state broadcast transmitters and vacuum tube transmitters.

The size of an all solid state transmitter approaches one-half the size of a tube transmitter of the same power level of recent design.

The signal quality of an all solid state transmitter can be perfected to exceed the transmitted quality of the conventional tube transmitter, especially important now that an emphasis is being placed on hi fidelity AM broadcasting as a prelude to stereophonic broadcasts by AM stations.

The reliability of an "all silicon" transmitter is enhanced because of the greatly reduced wear out characteristics of the transitor versus the vacuum tube. The tube becomes gassy, suffers from decreasing filament emission with age, and has a lower overall power conversion efficiency. The transistor arrays can provide a planned margin of power output capability in the event that a few transistors become inoperative. If the tube transmitter has but one final RF tube, the transmitter has no output when the final tube fails, but the solid state transmitter can maintain full or reduced output if some of the active output devices fail.

The economy of a solid state transmitter can be realized by its higher efficiency, longer life transistor active elements, and smaller space requirements.

The solid state transmitter design can easily implement power reduction without the complexity of high power contractors and power wasting dropping resistors, and a vernier output power control can provide an infinite number of reduced power output levels. By means of this feature, nonstandard operating power levels can be easily achieved, and instant on-the-air switching with no program interruption is possible.

Lightning and static discharge can be problems to any transmitter, but careful design can help to minimize possible damage and/or annoying program interruptions. Effective lightning and static protection can take the form of shunt static drain chokes, spark gaps, and reflectometer circuits.

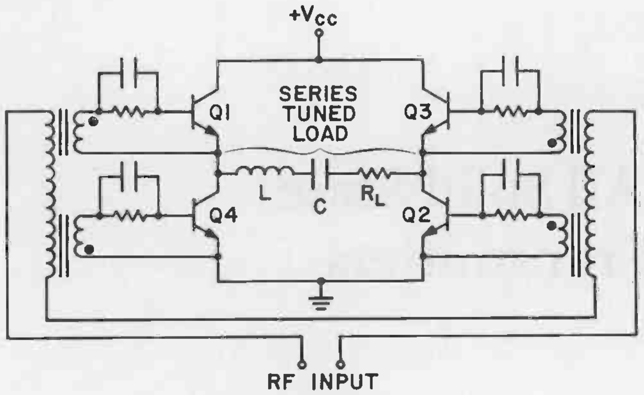


Figure 1.

In a solid state transmitter, the type of RF amplifier used can also enhance the protection of the overall system. The use of a push-pull bridge, saturated amplifier, commonly called a Class D amplifier, provides a sink to either the power supply or ground for any induced or transient energy.

The basic Class D RF bridge amplifier is shown in **Figure 1.**

Each arm of the bridge contains a transistor, and the RF input to the circuit is by means of an input transformer which has a single primary winding and four independent secondary windings.

The polarity of each secondary winding is such that transistors Q1 and Q2 are on and completely saturated for a given half cycle of RF while transistors Q3 and Q4 are turned completely off during the same half RF cycle. When the RF input reverses polarity during the next half cycle, transistors Q3 and Q4 are turned on and transistors Q1 and Q2 are turned off.

The time required to turn one set of transistors on and the other set off is extremely short—in the order of a few nanoseconds. During most of the RF cycle, the transistors are turned completely on in a saturated mode or completely cutoff, and the only time a small amount of power is lost in the transistors is during the nanosecond transition period and the saturation period.

The net result of the minimal power loss is excellent RF power amplifier efficiency in the range of 90 to 95 percent. If transistors were available which produced zero transition time and zero saturation voltage, the circuit conversion efficiency would be 100 percent, but the above mentioned circuit losses are ever present in the real world and limit the obtainable efficiency.

The RC network in the base circuit of each of the transistors produces a small amount of bias to help minimize the storage time effect of the transistors. The voltage produced across the series tuned load network is a square wave and the current through the load resistor, RL is a sinusoidal due to the filtering effect of the series network. The load resistor, therefore, has a sine wave of voltage across it and a sine wave of current through it.

The new generation of transmitters has reduced the operating controls to a minimum, and the familiar tune and load controls are no longer needed because these adjustments are present at the factory. The basic transmitter operating controls are "On-and-Off" and "Power Level Select."

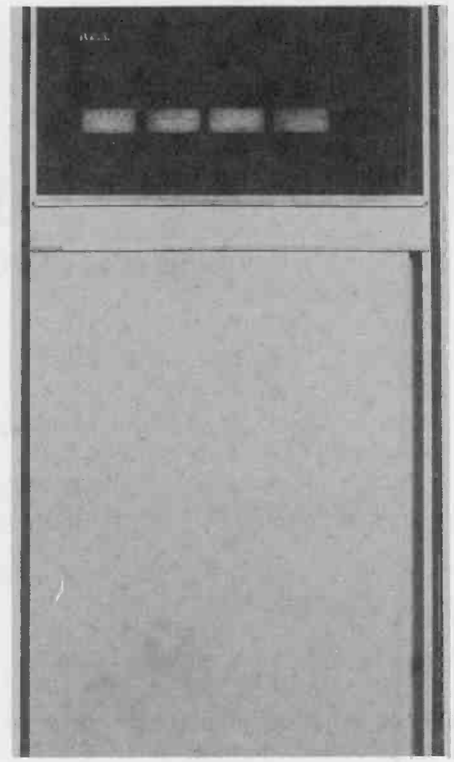


Figure 2.

BTA-5SS . . .

The first model in the RCA line of all solid state AM broadcast transmitters is the BTA-5SS. (See **Figures 2** through **5.**) This transmitter is a completely contained 5 kW carrier power transmitter which features low power consumption, high performance, and high reliability.

An overall block diagram of the BTA-5SS is shown in **Figure 6.**

The RF section consists of the following plug-in modules; the RF Generator, RF Pre-Driver, RF Driver and the RF Power Amplifier Trays.

The RF generator module **(Figure 7)** contains a high stability frequency synthesizer which allows the output frequency to be programmed in 1 kHz steps in order to satisfy both domestic and foreign frequency assignments. The heart of the synthesizer is a precise 5 MHz TCXO. The RF generator also has provisions for using an external frequency reference for synchronous stations and for frequency modulating the carrier for AM stereo applications.

The RF Pre-Driver module **(Figure 8)** is a buffer power amplifier between the RF Generator and the RF Driver Tray and is comprised of saturated Class D RF amplifiers.

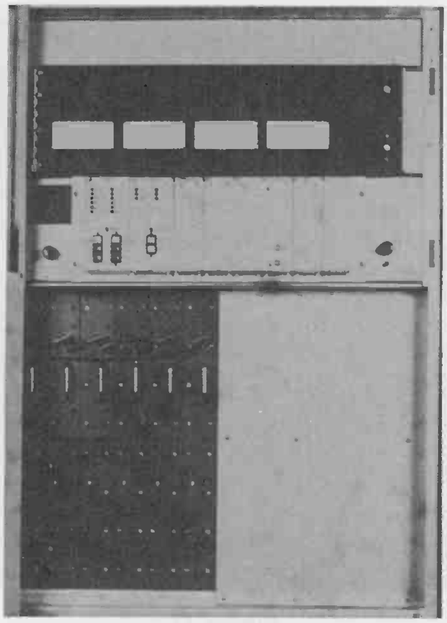


Figure 3.

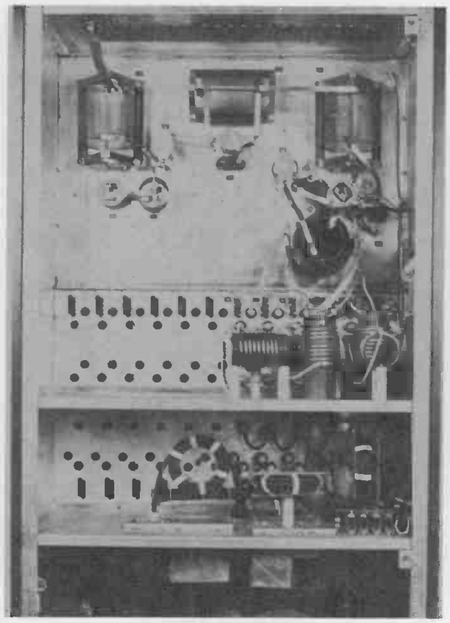
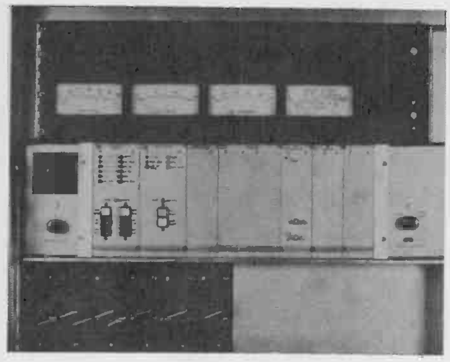


Figure 4.



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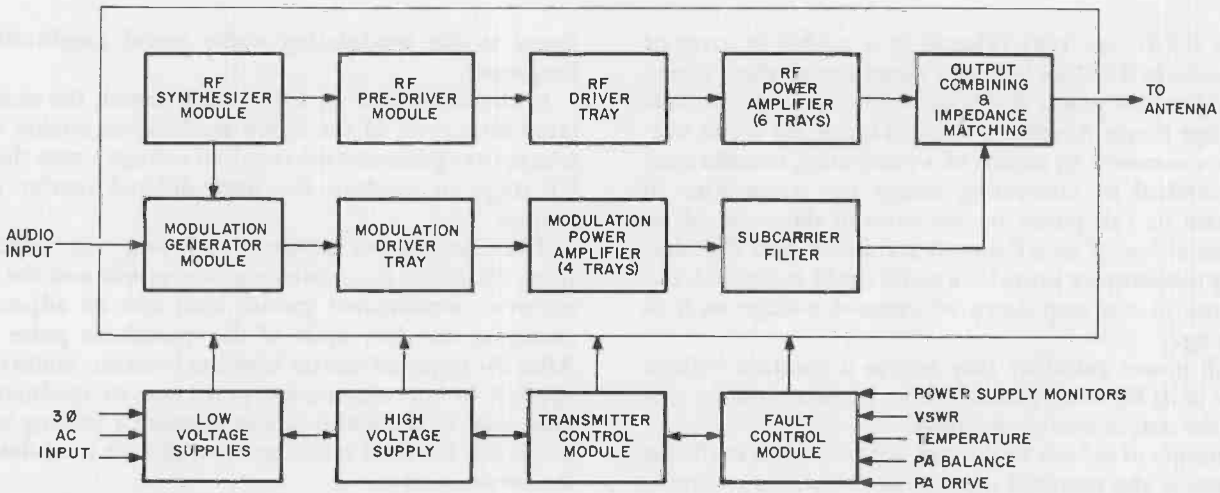

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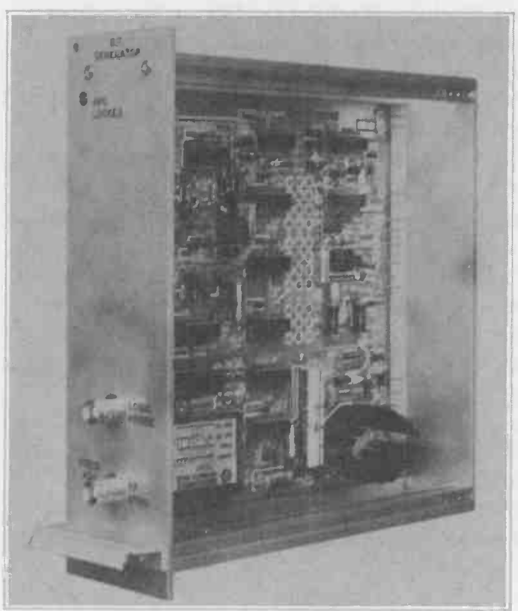
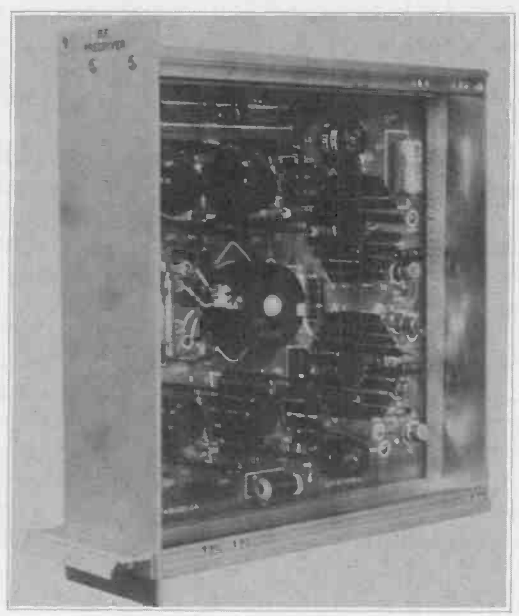


Figure 7.



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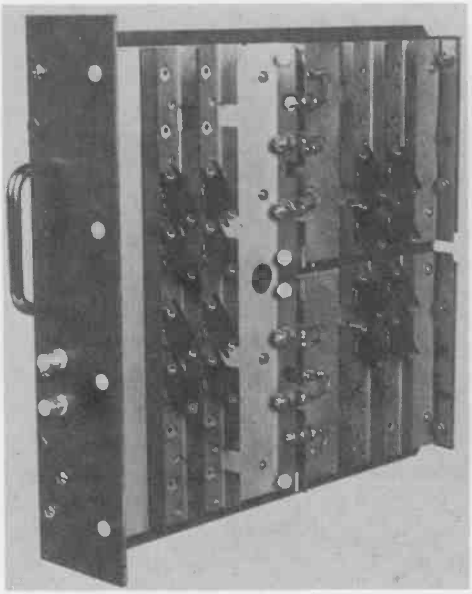


Figure 9.

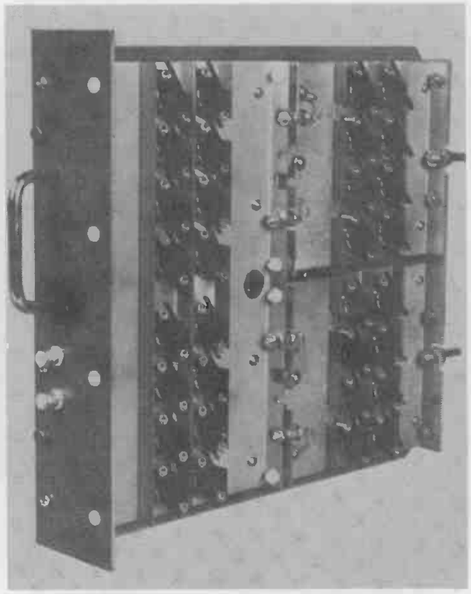


Figure 10.

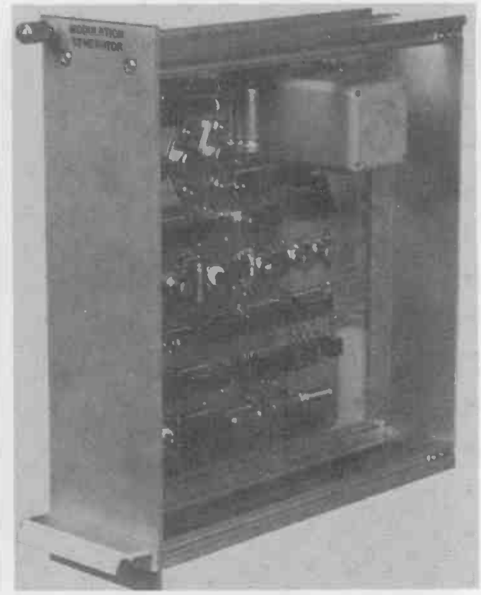


Figure 11.

The RF Driver Tray (**Figure 9**) is a plug-in array of transistors in the Class D Bridge Amplifier configuration.

The final RF power amplifier stage consists of six Class D Bridge Power Amplifier Trays (**Figure 10**) whose outputs are summed by means of a combining transformer. This method of combining allows the transmitter to maintain its full power output even in the event of an occasional loss of an RF output transistor, and the combining transformer provides a static drain to ground and a twenty to one step down of induced voltage such as lightning.

Each power amplifier tray acts as a constant voltage source to its RF load, and all of the transistors on the tray share the output current demand.

A margin of at least 25 percent is provided on each tray in terms of the required number of transistors to supply the required current output. This margin of safety means that at least 25 percent of the transistors on a tray would have to fail before the tray could not maintain its full output. An inoperative transistor is automatically removed from the circuit, and the remaining transistors continue to provide the full output current.

The final link between the combining transformer and the output to the antenna is the impedance matching and harmonic filter RF network. A reflectometer is also included to monitor forward power and VSWR and to provide protection by instantly quenching the RF output when transmission line disturbances occur.

The modulation system of the BTA-5SS utilizes a highly refined pulse width modulator. The subcarrier is directly derived from the frequency synthesizer in the RF generator module, and the resulting precise control of subcarrier frequency allows stable system performance.

The modulation generator module (**Figure 11**) produces a pulse train output with frequency equal to the subcarrier frequency and pulse width variations proportional to the modulating audio signal amplitude and frequency.

In the absence of an audio input signal, the unmodulated duty cycle of the entire modulation section of the transmitter generates the required voltage across the final RF stage to produce the unmodulated carrier power output.

The modulation section, including the subcarrier filter, functions as a variable power supply and the transmitter's unmodulated carrier level can be adjusted by changing the duty cycle of the modulator pulse train. After the required carrier level has been set, audio can be applied to the modulation generator to modulate the duty cycle at the audio rate to produce a varying voltage across the RF final resulting in amplitude modulation of the carrier output.

The entire modulator section consists of the Modulation Generator Module, Modulation Driver Tray, Modulator Power Amplifier Trays, and the Subcarrier Filter.

The Modulation Driver Tray (**Figure 12**) and the Modulator Power Amplifier Tray (**Figure 13**) consist of transistor arrays which turn on and off at the subcarrier frequency and in accordance with the modulated duty cycle.

The subcarrier filter removes the subcarrier frequency and applies a voltage, which varies at an audio rate, to the final RF amplifier. This modulation system provides low distortion, wide frequency response, fast transient response, high modulation levels, high efficiency, and a convenient method of adjusting and regulating carrier output power.

The Transmitter Control Module (**Figure 14**) and the Fault Control Module (**Figure 15**) provide complete control and protection for the transmitter.

The modules have remote control capability and a remote/local switch is provided for the safety of operating

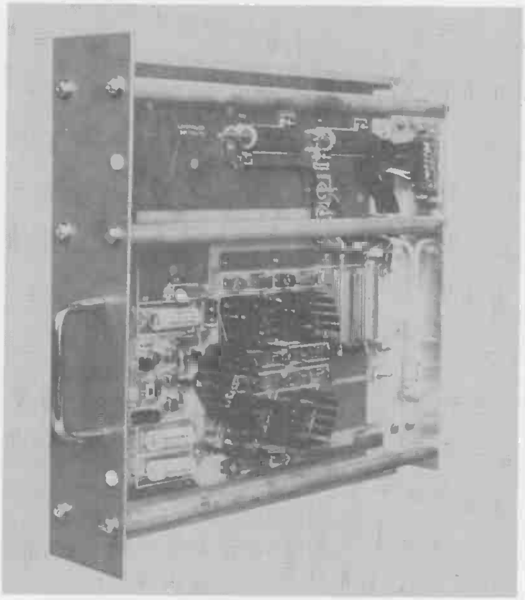
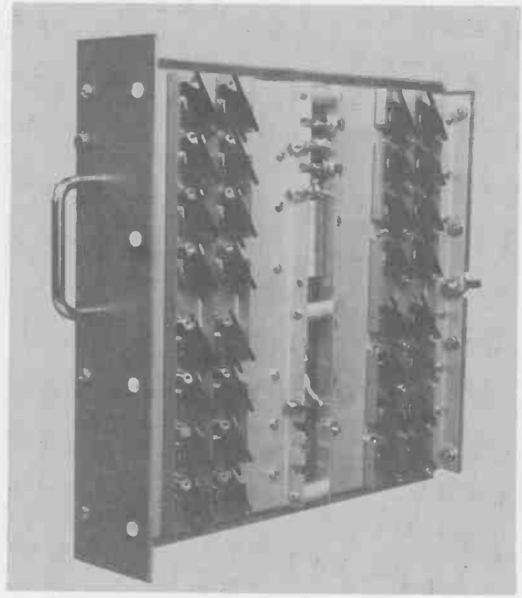


Figure 12.



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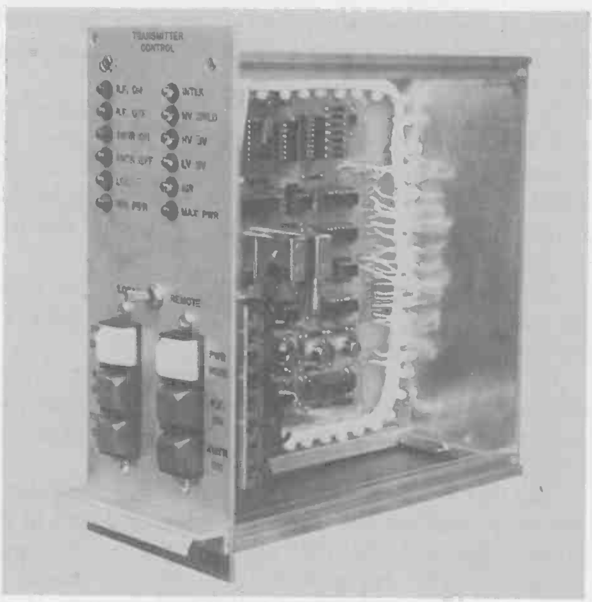

Figure 14.

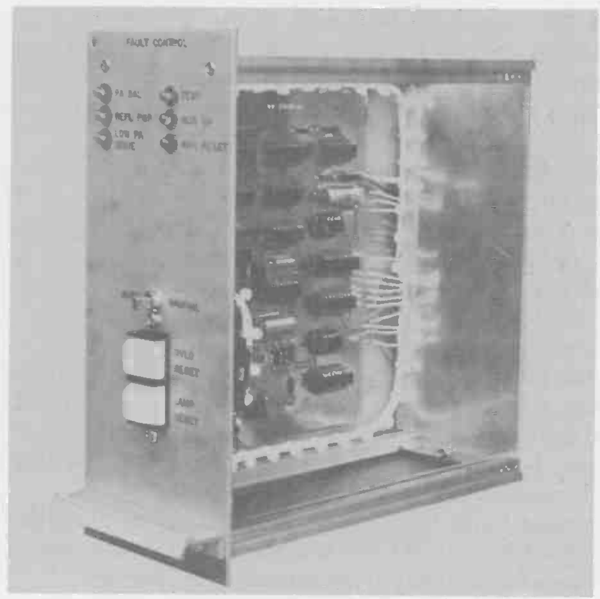

Figure 15.

personnel.

The main controls are: Transmitter On, Transmitter Off, RF On, and RF Off. A digital power increase/decrease control is also included and is controlled by two pushbuttons which give eight steps of power increase to 10% above nominal and eight steps of power decrease to 10% below nominal.

This digital power control increases or decreases the comparison voltage on the transmitter's automatic power control comparator. The power control comparator then adjusts the amplitude of the subcarrier triangle wave, and the resultant change of the triangle amplitude changes the duty cycle of the pulse width modulator.

As previously described, the transmitter's output power is adjusted by this change in duty cycle. A switch gives the operator the option of either automatic or manual overload recycle control, and a digital counter is provided in the automatic mode to set the number of overload steps allowed before the transmitter is shut down.

The high voltage supply is protected from overcurrent and undervoltage conditions, such as the loss of a single phase, and either condition shuts the transmitter down. The front panel indicators show the reason for shutdown.

The low voltage supplies are undervoltage protected and are current limited.

A reflectometer circuit sends a fault pulse to the control logic when a high VSWR condition exists and the transmitter's RF output is instantly cut off. The drive level to the RF power amplifier trays is monitored, and if inadequate drive is present, the transmitter protects itself by turning off.

The RF output level of each of the RF PA trays is detected by the tray balance circuit, and if the trays are not properly balanced in output, the transmitter does not allow operation until the tuning on the trays is set properly or the defective tray is repaired.

Under normal operation, the tray balance circuit provides a convenient check on tray performance.

The temperature of each of the RF power amplifier trays and the modulator power amplifier trays is monitored, and if a tray develops a higher than normal operating temperature due to a malfunction, the protection control circuitry turns off the system.

In the event of a failure of the blower, the air flow detector automatically reduces the transmitter output power and keeps the transmitter on the air. A front panel indicator is turned on when the transmitter is in this mode of operation.

The transmitter has four illuminated meters to monitor the RF PA Volts, RF PA Amperes, % Output Power/VSWR, and 20 circuit parameters on a multimeter.

The BTA-5SS offers the broadcaster high performance and economy. Here is a look at some of the transmitter's preliminary specifications:

A low distortion of 2% maximum from 30 to 10,000 hertz at a modulation depth of 95%.

A frequency response of 30 to 15,000 hertz which is flat to within ± 1.5 dB.

A high modulation capability of 125% positive peak modulation at an output power of 5.5 kW.

A low noise level of at least 60 dB below 100% modulation.

This design will result in a high volume, high fidelity AM broadcast signal, without sacrificing the designed overall system efficiency since the BTA-5SS provides an RF output to AC line input conversion efficiency of 65% or better at 5 kW output.

The carrier shift or carrier amplitude regulation of the transmitter is 1.5% or better at 100% modulation, and there is an automatic power control to maintain the transmitter's carrier output at a level which is preset by the broadcaster.

In addition, an automatic modulation control circuit will keep the modulation depth at a level preset by the broadcaster as the transmitter's power output is varied to eliminate the need to readjust the modulation level when a switch is made from high to low power or low to high power.

The automatic features of the BTA-5SS are designed to make the task of utilizing the FCC's Automatic Transmission System extremely simple.

In the design of a broadcast transmitter, it is very important to provide enough service features for routine inspection, cleaning, and in the event of a failure, easy repair.

The BTA-5SS makes use of extensive modular construction, and the low level nest modules shown in Figure 5 are designed so that they may be operated on a module extender. The transmitter cabinet was made large enough to give easy access to all components. A multimeter on the front panel gives operating parameters in 20 different circuits throughout the transmitter, and several illuminated status indicators are provided for instant evaluation of operational status or fault conditions.

Conclusion . . .

The age of all solid state medium and high power AM broadcast transmitters is here and offers the broadcaster high performance, economy, and reliability.

The future promises even higher power all solid state AM broadcast transmitters, and as the solid state technology advances, powers of greater than 5 kW will be possible.

It is expected that we will see a rapid increase in power output density in terms of watts per cubic foot of cabinet space. More self-monitoring and correcting features will be introduced as extensions of the present Automatic Transmissions Systems (ATS) are authorized by the Federal Communications Commission.

Automatic Transmission Systems For the Broadcast Service

John W. Reiser
Federal Communications Commission
Broadcast Bureau Reregulation Task Force

I appreciate the opportunity of discussing with you some of the details of the standards for Operation of Automatic Transmission Systems included in rules adopted in December by the Federal Communications Commission and some of the interesting questions and developments that have been brought to the commission staff's attention in the past several months.

Because of the limited time, it is inappropriate to dwell on the history of the ATS rulemaking proceeding, or to discuss in detail the basis for each of the technical provisions of the adopted rules.

I will outline briefly provisions of the rules and then report on some of the questions and comments we have received from licensees, broadcast engineers, and equipment manufacturers who are attempting to develop or use equipment that will truly provide beneficial operation and comply with the ATS standards as given in the rules.

It was certainly obvious from the onset of these proceedings that there was a considerable variation in concept, both within the industry and the commission staff, of what an ATS could or should be.

There is absolutely no doubt that electronics and broadcast technology is available to develop and operate a fully unattended automated transmitting Plant — and with program automation, a fully unattended broadcast station. That is evidenced by some of the equipment on display at this Convention.

Assuming that a highly sophisticated automated transmitting system can be developed, there still was the question of how much automation should be required, how much is necessary, or how much would be totally practical for a large number of broadcasters to use.

In the rulemaking process, whether it involves technical matters such as the standards for a directional antenna sampling system, or administrative or licensing issues, consideration must be given to the great variation which exist in the nearly 9,300 operating stations. A number of compromises and carefully balanced decisions are necessary in the process — hopefully successful but, unfortunately, not completely satisfactory to all.

Experience under adopted standards and rules indicate that revisions may be necessary or desirable. (One example of a technical rule which may require some revision is the standards for approval of directional antenna sampling systems. These standards do not permit the use of current transformers as sampling devices at the base of self supporter antenna towers. Some engineers tell us now that such transformers can be used effectively if certain precautions are taken, and that the rules should be amended accordingly.)

There has been some comment that the rules for ATS may not be practical for most broadcasters; that ATS offers no great advantage over the present remote control operation.

In this respect, I feel somewhat like the father of a newborn child . . . one who has looked forward to the birth of the child with great expectations, then, after the child arrives, wonders what the future holds for it. Will there be a fruitful life of joy and good fortune, or a life of desolate failure? Only time will tell, but constant encouragement, example, and correction when appropriate is required for the child — and perhaps also with ATS.

I would particularly like to point out that the adopted ATS rules are somewhat a noted departure from many rules ushering in previous technical changes in broadcasting.

During the past week while researching some rules of long standing, I was looking at the Regulations and Broadcast Engineering Standards published in 1939 . . . many of which are still in effect today. I was amazed at the very explicit detail that was included on the construction and installation of the transmitting equipment, including design considerations tube operating parameters, component specifications, sizes of conductors, thickness of insulation, etc.

In the ATS rules we attempted to avoid the inclusion of specification on the *design* of the ATS equipment. We tried to concentrate on the desired system performance.

The problem with rules that cover the specific equipment and design details is that they hinder the development and use of new technology. We hope that you, as broadcast engineers and equipment manufacturers, will devise effective equipment to meet station licensees' varied requirements.

Chairman (Richard E.) Wiley included the following in his statement on the adoption of the ATS rules:

> "In my opinion, the Commission's approval of ATS marks the beginning of what, hopefully, will be a new regulatory philosophy for the FCC, one which focuses more on the end product desired (in this case, the technical integrity of a station's signal) and less, much less, on complex rules and procedures to achieve that objective. Our action today permits licensees themselves, rather than government, to select the best means of maintaining the technical compliance of his or her particular circumstances."

What the Rules Provide . . .

Now, I would like to briefly outline provisions of the rules for ATS:

1. ATS is a technology whereby the routine transmitter operating functions can be performed automatically without operator supervision or control to insure transmissions are within specified standards and license conditions. These functions include making changes in the operating mode at the times required, maintaining operating power and modulation, and alerting the licensee of a condition that requires technical attention.

2. The use of an ATS relieves the licensee of the need to have a technician-type operator on duty responsible for observing the performance of the transmitting system and to make adjustments at prescribed times.

3. The use of ATS by any licensee is entirely optional. Licensee may use ATS for a portion of the broadcast day, and direct or authorized remote control at other times.

4. Type approval or type acceptance of ATS control equipment is not required. Licensees may either design their own equipment, have it custom designed or constructed, or obtain it from a broadcast equipment manufacturer.

5. No construction permit or prior authorization is required to install or test ATS equipment. After the licensee has completed the installation and testing to insure it is functioning correctly, an informal request to use ATS may be submitted to the FCC for authority for ATS operation. A detailed technical showing of the equipment installed is not required.

6. At the present time, all FM and AM stations when using non-directional antennas may use ATS. (It is anticipated that ATS rules for operating with AM directional antenna systems and TV stations will be adopted within the next nine month period . . . by early fall for AM directional operation, and by the year's end for TV stations.)

7. Upon receipt of the ATS authorization, the station can immediately begin ATS operations, during which one station employee must be on duty monitoring the transmission. Under the existing provisions of the Communications Act, that employee must hold a radio operator license or permit. A restricted radiotelephone operator permit obtained by mail registration is adequate. The monitoring operator is not restricted to a particular room or area; nor is he or she restricted in the other duties that can be performed. It is only necessary that that person monitor the station's transmission and an aural malfunction alarm signal.

8. The ATS equipment is to monitor and adjust the operating power by the direct method for AM stations, and by either the direct or indirect method for FM stations. The indirect method of power determination and maintenance can also be used at AM stations, under the same conditions — when necessary — such as when the antenna has been damaged or when antenna construction work is in progress.

9. The ATS must have a means for observing the depth of modulation, correcting excessively high modulation, and terminating the station transmission if modulation corrections are not made. Automatic correction of low modulation levels is desirable but not required.

10. For those AM stations operating only during daytime hours, under presunrise service authorizations or with more than one power mode, and stations sharing operating hours with other stations on a specified schedule, the ATS must include a time clock for performing all switching functions at the prescribed times. The clocks are to be accurate to within one minute of national bureau of standards time.

11. Each ATS station must have one or more monitoring alarm points at which the station employee monitoring the station is on duty. The points may be at the transmitter site, the authorized studio or remote control point, or at another location if specifically requested and authorized by the commission. The monitoring alarm point is to be equipped with an off-air monitor, SCA program monitor if SCA programming is transmitted, and an aural alarm signal that will indicate certain transmitter malfunctions. The signal must indicate an interruption in the station's transmissions — either carrier or modulation — for a period exceeding three minutes or uncorrected underpower operation. The monitoring alarm point must also be equipped with a means to turn the transmitter on and off.

12. The ATS is to be equipped with fail-safe devices that will terminate the station transmissions in event of certain serious malfunctions. These malfunctions include any uncorrected over power operation exceeding three minutes, uncorrected overmodulation, failure of the circuit permitting the transmitter to be turned off at the monitoring alarm point (a requirement similar to remote fail-safe requirements), failure of the mode switching time clock, and failure of the parameter sampling or alarm signal system.

13. The transmitter parameters to be automatically monitored may be checked continuously or sequentially at least once every minute. Modulation monitoring must be continuous.

Other Provisions . . .

There are several other ATS operating requirements or features for ATS:

1. The beginning of operation for each broadcast day or period is to be manually activated. For daytime stations, the time clock must prevent the station from signing on before the authorized time.

2. AM stations using a time clock for mode switching may have a switch to override the clock controls at the monitoring alarm point to transmit emergency information with full daytime power at any time.

3. Stations using ATS must comply fully with all emergency broadcast system (EBS) requirements.

4. The equipment must have some method for the maintenance operator to test the functioning of the automatic controls, fail-safe switching limits, and alarm circuits. It is not necessary that the transmitter actually leave the air during the testing. The testing thus can be done during regular broadcast operation.

5. All mode switching is to be accomplished without manual tuning or other adjustments.

6. The licensee may install additional alarm features in the ATS equipment. There must be a distinction, however, between required and optional alarms.

7. The use of auxilliary or alternate main transmitters can be incorporated into the ATS system.

8. The ATS automatic alarm monitoring for tower lighting failures is optional. Licensee can continue to make daily observations of the tower lighting, either directly or by remote telemetry, if desired.

9. There were no changes in transmitting equipment or signal transmission standards in the ATS rules. All stations using ATS must meet the same power and modulation standards as those not using ATS. However, modulation peaks of frequent recurrance is specifically defined for ATS control purposes.

Some ATS Benefits . . .

As mentioned previously, questions have been raised as to whether there would be any interest in ATS, and what advantages exist under the rules as adopted.

The ATS proceeding certainly generated considerable comment and supporting interest from broadcast licensees.

A number of manufacturers have developed equipment for marketing.

I am aware of one licensee that has already been authorized to use ATS, and others have told us that they are now building, installing, or testing ATS equipment.

In the long run, I am sure you realize that the acceptability of ATS will rest in the economic or operational advantages that it provides. — hopefully both advantages. And we also trust that it will provide a more reliable quality service to the public.

During the ATS rulemaking proceeding, we were neither given information, nor could we possibly develop any prediction, on the economic advantage that stations may realize by using ATS. This would be almost entirely dependant upon a station's existing facilities, its type of operation, location, staff resources, cost of ATS equipment, and a number of other factors.

I suggest a few of the benefits of using ATS:

1. Licensees may have employees on duty who are no longer required to take and pass a written operator license examination. Licensees have greater flexibility in the selection of station operating staff and staff duties.

2. The station employees on duty during station operation are relieved of all technical responsibilities for transmitter operations or of being restricted to a particular room or operating position. For example, the duty employee could be an announcer, receptionist, switchboard operator, or watchman.

3. Licensees will be relieved, we assume, from their expressed anxiety concerning the reliability of operators in performing such duties as switching power modes on time (a problem experienced with many "combo" announcer-operators).

4. ATS can provide a more reliable service to the public, including the lessening of potential of interference to other stations caused by improper attention to mode switching, or excessive modulation.

5. The alarm system will call attention immediately to a technical malfunction rather than the malfunction detection being dependent upon the operator's observations.

6. There will be a reduction in the number of inspections that must be made at the transmitter site. ATS stations may inspect the transmission systems on a monthly schedule.

7. Stations using ATS may employ the services of a first class operator on a contract or part time basis.

8. We have for the first time in the 33-year history of the FCC a standard for that previously ambiguious phrase "peaks of frequent recurrance" with respect to maintaining modulation levels.

9. Routine transmitter meter readings and logging of operating parameters are not required with ATS.

Objectives . . .

It has been mentioned that many of the technical devices required for ATS operation could have been, or may be, used by any station to achieve many of the advantages listed above without ATS.

The equipment may have been easily and economically built and used . . . but it wasn't. Many stations, even under the existing operator logging and inspection requirements, failed to comply with the most basic technical responsibilities, such as switching operating modes at the prescribed hours.

We trust that the ATS rules will be successful in providing the regulatory basis for improved station operations. There are a number of methods by which the requirements for an ATS can be accomplished, and a number of excellent systems have been, and probably will be, developed.

This equipment also surely will be used to improve the operation of those stations that do not elect to implement full ATS operation.

Some Questions and Answers . . .

I would like now to review some of the questions and comments that have been received since January concerning the ATS rules and requirements.

Q. Is it necessary that the ATS system actually adjust the output power of the transmitter, even if alarms are provided for over or under power?

A. Yes. One purpose of ATS is to keep the station operating as close to the licensed power as possible without operator involvement. This is to be accomplished by automatic adjustment of the transmitter operating power.

Q. Can a transmission line sampling or in-line watt meter be used to maintain the output power of an FM transmitter if the sampling device is calibrated using the indirect method of power measurement?

A. No. A transmission line power sampling device must be calibrated with a wattmeter using an artifical antenna. Or, the indirect method of determination and maintenance of the operating power of FM transmitters (the product of the power amplifier voltage, and current, and efficiency factor) is to be used. It may be desirable to make some revisions in the rules covering the determination and maintenance of operating power for all stations to permit the use of other methods.

Q. If a microwave STL is used to transmit composite stereophonic signals from the studio to the transmitter, can the modulation monitoring and control function be located at the studio where the stereo generator is located?

A. Yes. The ATS rules do not specify where the particular components of the system must be located. From FM, an off-air monitor, if used with a receiver or RF amplifier or adequate bandwidth, can give an accurate representation of the depth of modulation. Off-air monitoring for modulation control may not be practical for AM stations unless the monitoring point is very near the transmitter. Atmospheric noise, and co-channel or adjacent channel interference will affect the accuracy of off-air AM modulation measurements.

Q. Do stations using ATS require a modulation monitor, and if so where is it to be located? Is it necessary that the modulation monitor be in continuous operation?

A. All stations are still required to have a type approved modulation monitor. But it need not be in continuous operation. It can be located at the transmitter site for test and maintenance purposes, or at the monitoring and alarm point and used as an off-air monitor receiver.

Q. The ATS standard for a burst of modulation differs from the specifications for type approval of modulation monitors. Why?

A. The specifications for AM monitors, FM monitors used for monaural signals, and stereophonic modulation monitors all have some variations. It was not possible to adopt the existing monitor specifications to ATS operation so a new basis was established for defining excessive peaks of modulation for ATS control.

Q. Can existing modulation monitors be used for ATS purposes?

A. Yes, if they can be adapted for such use. The ATS logic will probably be external to the monitor signal terminals. Caution is required to insure that the modulation indication is accurate for each AM station power mode used, and the accuracy of an ATS modulation peak sensing device must be immune to possible variations in the RF input level.

Q. Can FM SCA or subaudible tones on AM stations be used for transmitting alarm functions to the monitoring alarm point?

A. Yes. The SCA FM Rules or the AM rules were not specifically amended to provide for ATS alarm or control signals, but rule changes will be included if necessary in the second report and order for such use.

Q. Is it necessary that the ATS alarm circuit be self-testing at least once every minute, or that subaudible tones be transmitted continuously to indicate that the alarm circuit is not functioning?

A. The ATS fail-safe system must have a provision for determining that each component of the alarm function is operating — the sensors at the transmitter, the link from the transmitter to monitoring alarm point, and the alarm circuitry. If a wireline is used to transmit the alarm from the transmitter site, then an interruption of that line should result in a turn off of the transmitter. If AM off-air monitoring is used for the link, a continuous tone would not be needed assuming that either an alarm failure at the transmitter site or at the monitoring and alarm point would turn the transmitter off.

Q. Is it necessary that the ATS system provide for automatic alarming of tower lighting failures, and, if so, must it indicate if any single lamp fails?

A. Automatic alarm for tower light failure is not required for ATS operation, although it would be desirable. If such alarming is used, it must indicate if there is any failure of a lamp. The licensee would be obligated to determine which lamp has failed so FAA can be notified if necessary. Note that FAA notification is required only if the topmost lights on the tower or any flashing code beacon have failed. If an automatic lighting failure alarm is not used, licensees may use either direct observation or telemetry checks or tower lighting each day and a log must be kept of daily tower lighting observations.

Q. Why was there no provision in the rules for AM stations determining power by the indirect method?

A. The report and order stated that the indirect method of power determination could be used with ATS under the same temporary conditions as permitted under present rules. This was not included in the ATS AM rules through oversight and will be corrected in an order to be released soon.

Q. Can FM stations that have a combined compressor limiter-stero generator meet the ATS modulation control requirements?

A. Some modifications of the combination unit may be necessary, and it is suggested that licensees using such devices contact the equipment manufacturer. No station was prohibited from using ATS because of the type of equipment now in use, and modifications are permitted so long as signal transmission standards are maintained.

Q. The proposed ATS rules stated that control time clocks required presetting for at least one month in advance, but the adopted rules did not require this. What happens if a station forgets to adjust the clock for the first day of the month?

A. A clock for presetting for the following month was not required because we believe the licensee should have greater option in selecting ATS equipment for a particular station operations. If the licensee forgets to have the clock adjusted for the first day of the following month, he or she would be subject to the same penalties for operating at variance with the license terms as under manual operation. If experience shows that there is a general problem with stations not resetting the clocks when necesary, some modification of the rule may be necessary. I understand that one manufacturer is using a clock calendar integrated circuit to provide preset switching for the entire year with a reserve power supply to keep the clock running during a power failure.

Q. Why was not automatic frequency control or fail-safe for off frequency operation made part of the ATS rules?

A. Off frequency is such an infrequent problem in broadcasting, that the additional equipment required for an ATS frequency monitor is not justified. The monthly frequency measurements are usually sufficient to detect any abnormal frequency drift. Some older stations using crystal ovens may be off frequency after a long period of power interruption, but frequency monitoring equipment is also subject to similar problems. All licensees using transmitters that could be off frequency after extended power interruptions should take steps to insure that the frequency is within tolerance when resuming operation.

Q. If a station goes off the air because of some interruption at the transmitter, can it return to the air by ATS control?

A. Many transmitters have the ability to recycle after brief power interruptions, lightning strikes, or overloads. This automatic recycling is normal broadcast procedure and is not affected by the ATS rules. However, there could be a situation where there are longer interruptions in the power service to the transmitter. Since the ATS alarm must activate for a transmission interruption lasting longer than three minutes, we think that it is reasonable to have manual transmitter turn on if the interruption lasts longer than three minutes.

What Lies Ahead

Looking very briefly to the future of ATS, I see no reason why AM stations with directional antenna systems can not use ATS.

The few problems remaining to be resolved concern use of existing antenna monitors, interval of parameter samplings, reasonable limits for directional antenna parameter deviations before the station must actually terminate operations, and procedures for use of ATS when there is damage to the antenna, or when modifications are in progress.

Similarly, we expect that TV stations will also be able to use ATS, and our attention will be directed to those signal parameters which must be automatically corrected, the parameters that must be alarmed for manual correction, and the appropriate video test signal necessary to accomplish ATS monitoring and control of the visual transmission.

TV Frame Synchronizer Applications

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The TV Frame Synchronizer, although a relatively new device in broadcast equipment product lines, has already become familiar to most of us.

There have been several papers recently introducing synchronizers. These papers have usually dealt with the basic concept of a synchronizer and have described preferred system implementations. However, they have not, in general, discussed the synchronizer's internal operations.

This paper will take a closer look at the operation of a synchronizer, especially the memory. After examining the basic operation, it will then be possible to look at present options and describe what happens in the memory to make these special effects possible.

Figure 1 is a typical block diagram of a Synchronizer.

Three video signals are shown—the Remote Video Input, a Studio Color Video Reference and a Synchronized Video Output. The Remote Input and the Synchronized

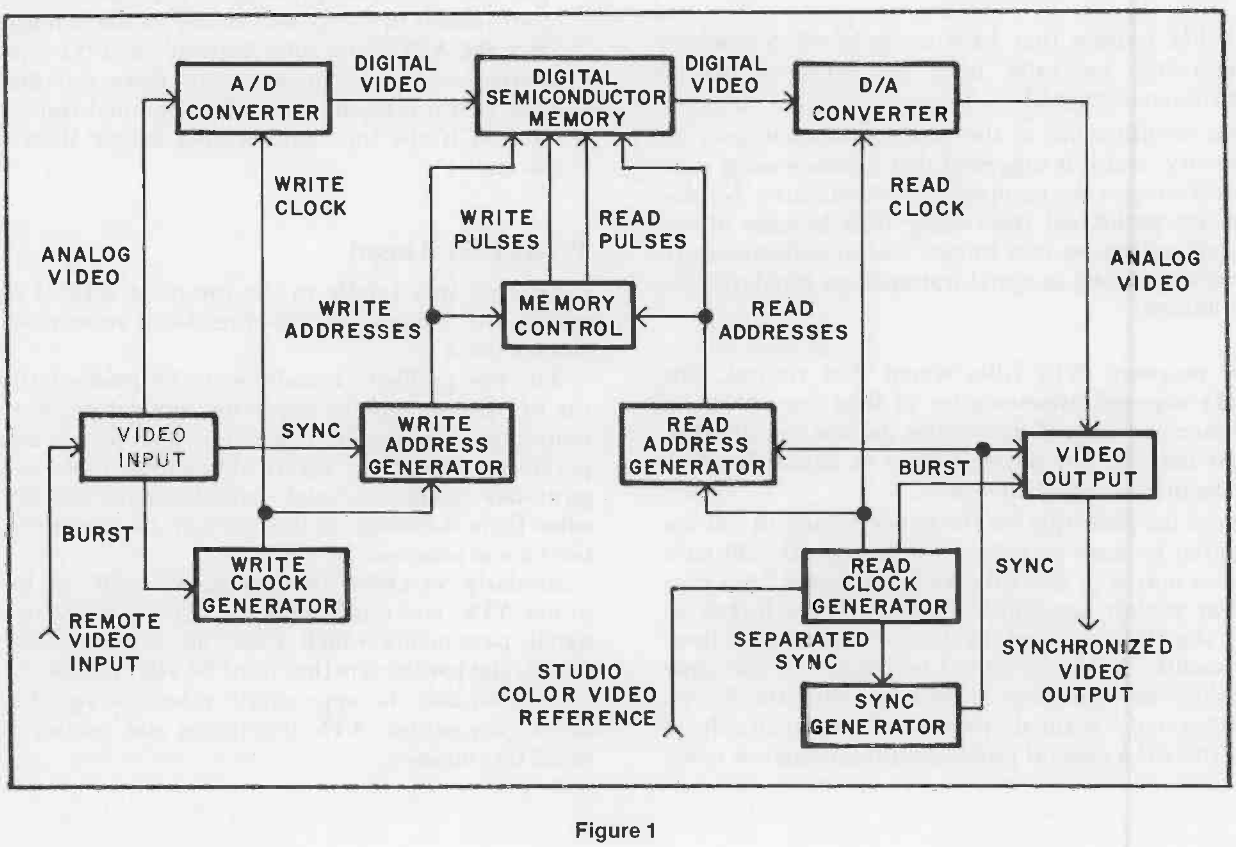

Figure 1

Output are identical except that the output is timed precisely with the reference rather than with the input.

The Synchronizer has been described as a variable delay line where the delay is precisely that which is necessary to phase the output horizontally and vertically (including subcarrier phase) with the reference. This is accomplished by writing the input video in a memory and, after the proper delay, reading the video out of the memory.

How It Works . . .

The remote video signal is received by an Input Video Processor. The primary functions of the processor are to clamp the analog video prior to being converted into a digital signal by the A/D Converter and to extract sync and burst from the video signal.

The extracted burst is presented to the Write Clock Generator whose function is to provide a series of sampling pulses to the A/D Converter for digitizing the video signal. The extracted sync is delivered after processing to the Write Address Generator, enabling that circuit to generate unique addresses for storage of the digital video in the memory.

The studio video reference is received by a Read Clock Generator whose function is to extract sync and burst from the reference video. The extracted sync is delivered to the Sync Generator which gen-locks to the reference video and delivers processed sync to the Read Address Generator.

The Read Address Generator causes digital video to be read from the memory by producing the same sequence of addresses that was generated by the Write Address Generator.

The Read Clock Generator uses the extracted burst to generate a series of re-sampling pulses which are delivered to the D/A Converter for purposes of converting the digital video signal back into an analog video signal.

The Output Video Processor accepts this analog video signal, inserts proper levels of sync and burst, and delivers the processed signal to the output terminals of the Synchronizer.

The Memory Control is responsible for looking at the write addresses and generating a write pulse at the proper time. The digital video arriving from the A/D Converter thus is stored in the memory at the specified address.

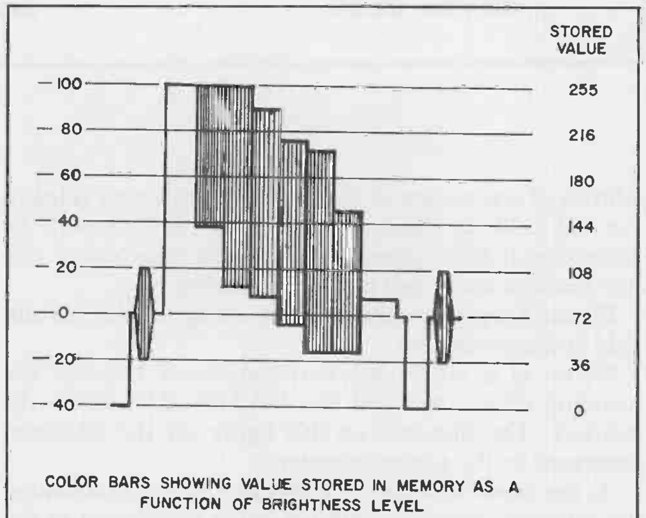

Figure 2

Memory Operation . . .

One frame of the video signal in the memory is composed of 393,216 picture elements stored as a discrete number.

The discrete number refers to the brightness level. For example, the discrete value of zero would be the blackest video encountered and the discrete value of 255 would be the whitest video encountered. All other numbers refer to some grey level.

This is illustrated in **Figure 2** with color bars.

To be able to store these 256 different values requires 8 bits of memory for each and every picture element. The entire memory then requires 3,145,728 bits of storage.

The Memory Control likewise is responsible for looking at the read addresses and generating a read pulse at the proper time, causing the digital video to be read from the memory at the specified address and then delivered to the D/A Converter.

The addresses given by the Write Address Generator specify the location in the memory into which each picture element will be placed. To facilitate understanding of the address scheme, assume the address generators count from one to 393,216 in one frame, and that the

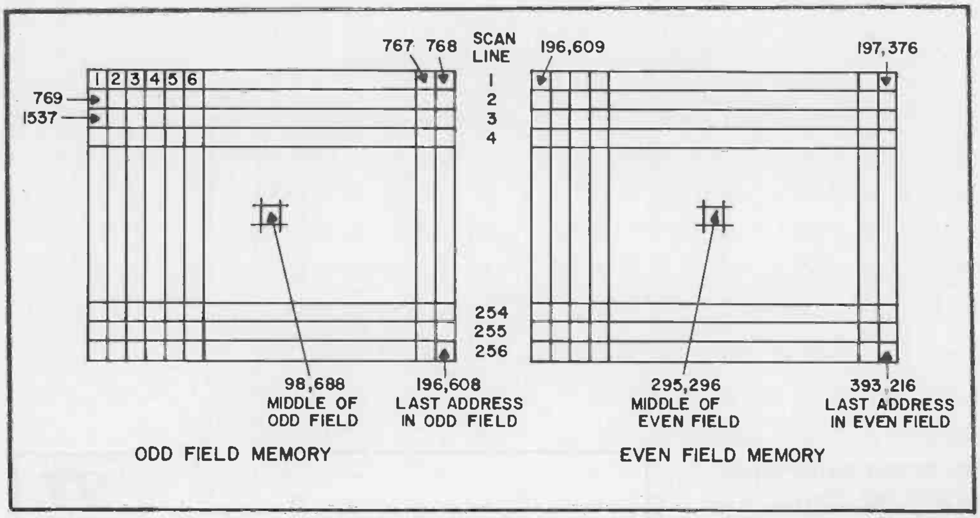

Figure 3

Correspondence between scanning raster and memory addresses for both fields. Each small box stores one picture element.

address of one occurs at the beginning of active video in the odd field. In this way, the memory is "scanned" by using digital IC counters in exactly the same way a picture monitor is scanned by an electron beam.

Figure 3 represents the memory storage of both an odd field and an even field.

Shown is a one-to-one correspondence between the scanning of a raster and the 393,216 addresses in the memory. The numbers on this figure are the addresses generated by the address counter.

As the input video scans the raster, it is also scanning the memory—except a number value is assigned to the brightness level of the video and that number is stored in the memory just as a number value is stored in a digital computer. Synchronization can then occur, because the stored numbers can be read from the memory after the necessary delay time has elapsed.

The Read Address Generator makes addresses for reading the digital video in precisely the same way the Write Address Generator made the storage addresses—except these addresses are referred to the sync of the studio reference rather than the remote video.

Picture Freeze . . .

Once this frame of storage is available, there are other things besides synchronizing that can be done.

For example, to freeze a picture it is only necessary to terminate the storage of new video in the memory.

This is done by eliminating the write pulses from Memory Control that were forcing the memory to store new video. The read circuitry continues to generate read addresses and read pulses. As a result, the output video will be the stored video repeated over and over until storage of the input video resumes.

The stored or frozen picture will not deteriorate with time because the semiconductor memory can hold the stored numbers as long as desired in the same way a computer holds stored numbers.

There is one detail that should be mentioned. When continually reading the stored numbers from memory

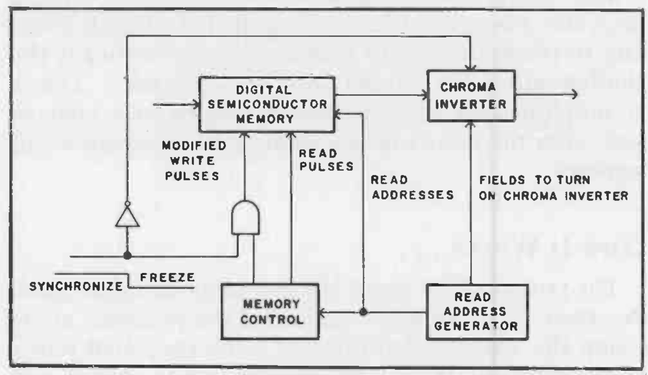

Figure 4

during a freeze, because of the frame-to-frame color subcarrier phase differences, it is necessary to use a chroma inverter to have proper color phase.

Figure 4 illustrates the necessary modifications of the Synchronizer block diagram to accomplish picture freeze.

A gate is used to interrupt the write pulses and at the same time turn on the Chroma Inverter during the fields designated by the Read Address Generator.

The Chroma Inverter could be either a digital chroma inverter or an analog chroma inverter.

Clean Up . . .

A similar application of a Synchronizer is to clean up any non-synchronous switches of the input video.

The typical Synchronizer will have circuitry which constantly monitors the input video. If the sync of the input video suffers a sudden unexpected change, the Memory Control write pulses can be eliminated just as they were for picture freeze.

Once the input video circuits have been able to genlock to the new input, the write pulses will resume storing the digital video in the memory at the next vertical interval.

During this time interval, the read pulses will continue to read the digital video held in the memory. As a result, there is a synchronous vertical interval switch at the out-

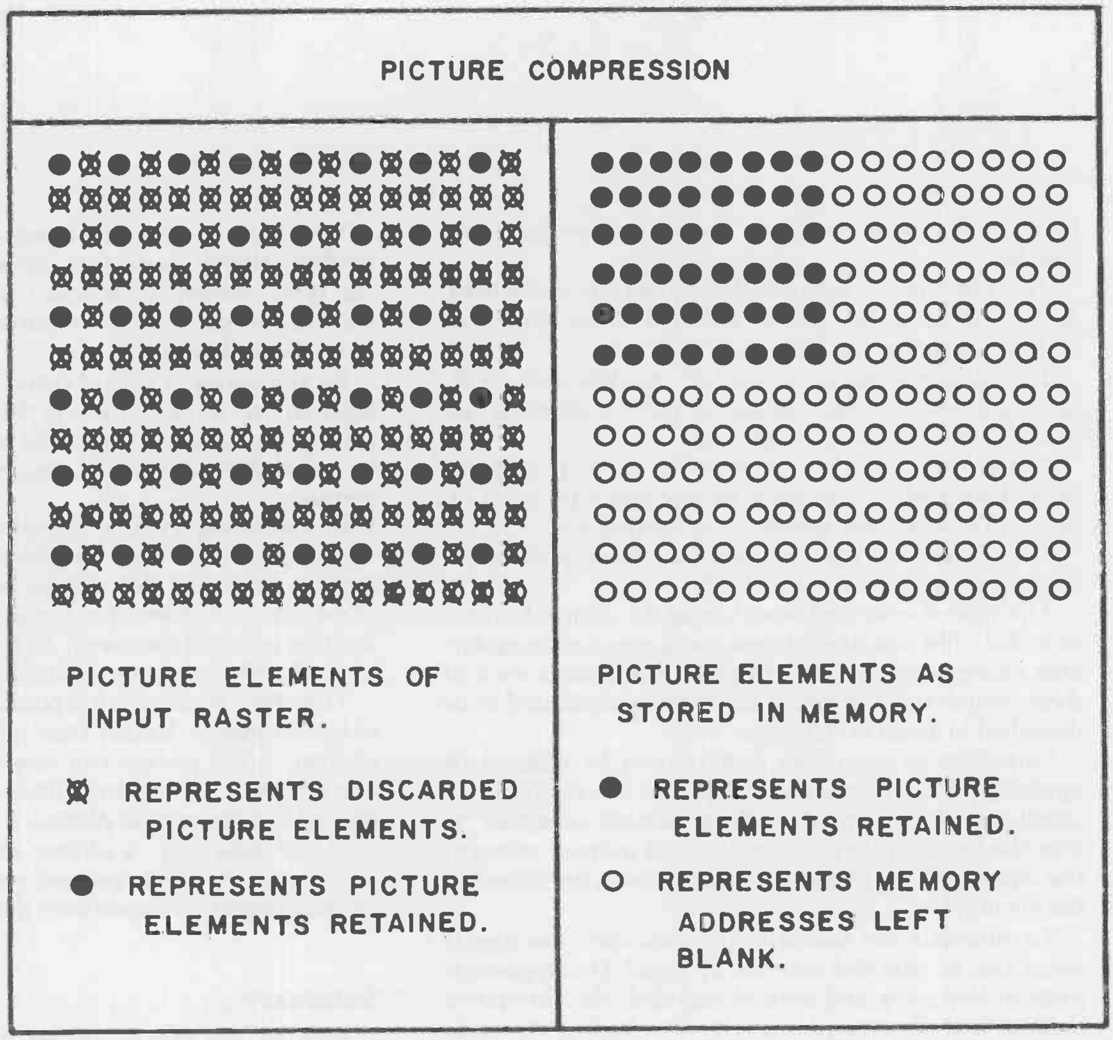

Figure 5

Some picture elements are retained as shown at left. The retained picture elements are placed next to each other in the memory as shown at right.

put of the Synchronizer even though there was a non-synchronous switch at the input of the Synchronizer.

By detecting whether the non-synchronous switch occurred during the storage of an odd or even field and forcing the read addresses to specify only the opposite field, there will be no visible tears in the output video.

The Synchronizer can even perform as a super drop out compensator as a result of its ability to handle non-synchronous switches.

If the input video totally disappears the write pulses will again be eliminated causing the Synchronizer to produce a frozen picture. When the input is re-established, the frozen picture will disappear and the live picture will continue.

A situation like this occurred during the televising of the "Great American Celebration" from Baltimore when a microwave feed from San Diego was temporarily lost. Because a Synchronizer was used at the receiving end, there were no breakups in the picture even though the input was totally missing for a moment.

Because sync and subcarrier are referenced to the studio and because they are properly established at the output of the Synchronizer, the only effect of a loss of input is the frozen image seen at the Synchronizer output.

Picture Compression . . .

Picture compression is another special effect which can be done with a Synchronizer.

A simple way of explaining the technique of picture compression can be seen by again examining Figure 3.

Suppose every other picture element produced by the A/D Converter during the first scan line of the odd field is literally thrown away and the remaining picture elements are placed next to one another in the memory.

This is shown in Figure 5.

For example, the first picture element is placed in address one, the second picture element is discarded, the third picture element is placed in address two, the fourth picture element is discarded, the fifth picture element is placed in address three, etc.

Note that the first scan line of the input video will be located in the first half of the first line of the memory.

Suppose then that the second scan line of the input video is completely discarded. The third scan line of the input video is then placed in the memory in a manner identical to that of the first scan line and in the memory locations that would normally be occupied by the second scan line.

This procedure is followed throughout the entire field. As a result, a smaller picture is stored in the upper half and the left half of the memory.

The picture elements at the left, marked with an X, are discarded and the remaining picture elements are stored as shown in Memory at right.

If this stored data is read from the memory with the normal method, the original picture will have been reduced to precisely one quarter of its normal size.

The technique, however, does have some problems if implemented in the way described.

This type of reduction would cause the color subcarrier to be lost. The resulting picture could also have considerable moire patterns. Procedures used to correct each of these undesirable effects are much too complicated to be described in detail in this paper.

Correction of the moire patterns can be effected by applying digital filtering techniques to the digital video signal. Rather than discarding picture elements as described earlier, they are used to find average values of the video and the resulting average values are stored in the memory.

To eliminate the loss of color subcarrier, the digital video can be decoded into its Y, I and Q components prior to averaging and then re-encoded. An alternative method is to average picture elements having the same phase of subcarrier. In this latter case, the subcarrier is automatically retained.

Picture Positioning . . .

Referring again to Figure 3, the relative ease of moving a picture around the TV raster, and even completely off the raster, can be seen.

Normally, video of the odd field is stored with the top edge of the picture in addresses one through 768. Likewise the left edge is stored in addresses one, 769, etc. In other words, the top left corner of the picture is stored at the top left corner of the memory.

The top left corner of the picture could as easily have been stored at the center of the memory. In this case the picture element normally stored in address one is stored in address 98,688, the center of the memory. The picture element normally stored in address two is stored in address 98,689, etc.

This would cause the top left quarter of the picture to appear in the lower right quarter of the output video picture.

In a similar manner the top left corner of the picture can be stored at any point in the memory. Or, the top left corner could be moved off the top of the raster, or off the left of the raster, or any combination.

By using a conventional positioner to specify the desired location of the picture, the normal address given by the Write Address Generator can be modified in such a way that the picture can be moved around in the memory to any desired location.

By generating a keying signal timed with the picture location and using this key as the external key input to a production switcher, any other synchronous picture can be inserted into the area vacated by the Synchronizer picture.

In describing Picture Compression, the first picture element of the odd field was placed in address one. By using the positioner this address can also be modified to cause the compressed picture to appear at any desired location on or off the raster. In this case the keying signal is timed with the picture location and size.

This effect is one which is possible only since the advent of Synchronizers. Rather than wiping from one signal to another, a full picture can now be moved off-screen in any direction un-masking another picture that was hidden behind the original picture.

In the same way, a picture can be moved from off-screen over top of the original picture. This new picture can be brought on-screen from any desired location.

Summary . . .

Each of the effects described was accomplished by modifying the normal sequence of writing the digital video into the memory.

For picture freeze, the writing of digital video was stopped.

For Picture Compression, the digital video was averaged and some picture elements were deleted before writing data into the memory.

For Picture Positioning, the write addresses were modified with a positioner.

In each of these cases, reading data from the memory was not affected except for the use of a chroma inverter whenever data storage in the memory was stopped.

In the short time that synchronizers have been with us, we have already seen great changes occur. The latest synchronizers are one-tenth the size, one-tenth the weight, and consume one-quarter the power of the earliest synchronizers.

The synchronizer originally was made possible by accomplishments in digital integrated circuit technology. As that technology has advanced, it has made possible the great changes we have seen in synchronizers.

As video engineers have become more familiar with these integrated circuits they have been able to design a variety of effects that were not available with the first synchronizers. The marriage of television and computers has produced today's digital video synchronizers.

A New Approach To Modulation Control

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In recent years great emphasis has been placed on sustaining high modulation levels in AM broadcasting. A great variety of compression amplifiers, limiting amplifiers, and asymmetrical audio processing equipment has been developed and presently is in use to accomplish this end.

While most devices now in use are remarkably successful, they process the audio material only and do not directly make use of the modulation characteristics of the final transmitted signal. They alter, at least to some degree, the dynamic range and other qualities of the program material.

It has been found by many broadcast engineers that when adjustments are made to obtain a very high modulation level with these devices, variations in the transmitter characteristics can cause overmodulation. If the adjustments are made to avoid overmodulation in the worst case, reduced modulation levels must be tolerated on the average.

The equipment to be described here is intended to close the control loop around the transmitter in order to correct for these variations so that high modulation levels can be maintained for all operating conditions.

We call this device an Amplitude Modulation Controller.

Basically, it samples the RF leaving the transmitter, makes measurements of the modulation characteristics, and by a digital logic process adjusts the audio level to the transmitter. It can be used in conjunction with all of the existing program processing equipment, and field tests have shown that under normal conditions a further enhancement of the modulation characteristics can be obtained.

The audio control on this modulation controller is strictly linear so that no real time compression or asymmetry is added to the program.

The Equipment . . . and How it Works . . .

The simplified block diagram in **Figure 1** will illustrate the operating principles of the Amplitude Modulation Controller.

A sample of the transmitter output is taken, typically with a toroidal current transformer, and supplied to the device.

This sample is first demodulated by an envelope detector. The output of this detector contains a dc level proportional to the unmodulated carrier and a superimposed audio component from the amplitude modulation.

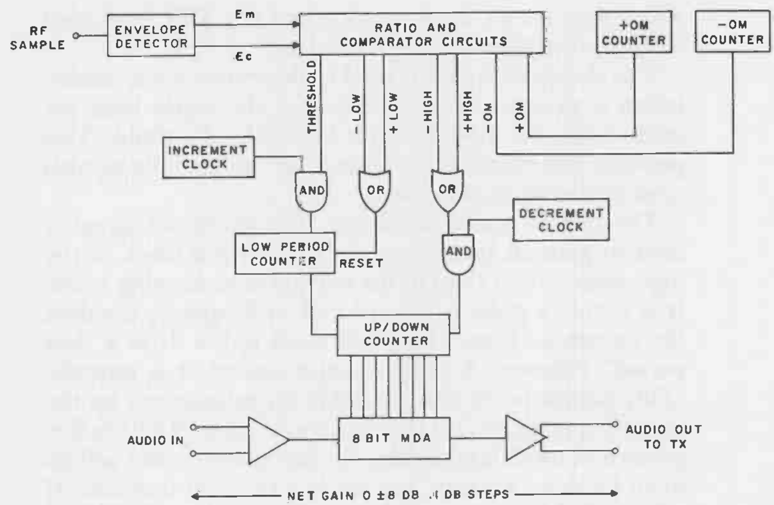

Figure 1.

This total envelope output signal is identified as E_M in the illustration.

The second signal, E_C, is derived with a unity-gain, low-pass filter. The result is a dc voltage proportional to the carrier level.

From the ratio of these two signals, real time modulation characteristics are determined. Typically, one of the signals is attenuated by a variable resistance divider and compared to the other with a digital comparator circuit.

For example, the comparison marked "Threshold" in the illustration is derived by taking 75% of E_C and comparing it to E_M. When E_M goes lower than the attenuated carrier sample, the comparator output goes high and indicates that a modulation in the negative direction exceeds 25%. A total of seven such comparisons are made simultaneously.

In addition to the threshold comparison just described, negative low modulation and positive low modulation comparisons are made. These levels are typically minus 85% and plus 100%. The plus and minus high modulation comparisons are set by the FCC limits, minus 100% and plus 125%.

The figures just given are typical only and the station operator is at liberty to set all of these levels at any desired values. But I will use the typical values throughout to simplify the discussion.

The comparator output signals are TTL logic level, that is, they are at a TTL low level when the condition for

Figure 2.

which they are set is not satisfied and at a TTL high level when the condition has been satisfied.

The threshold signal is used to determine when modulation is present. No corrections of the audio level are made when the modulation is below the threshold. This prevents the circuit from "pumping" as is common with most gain control amplifiers.

The circuit works as follows. The threshold signal is used to gate an increment clock. This is a clock in the logic sense rather than in the normal time-keeping sense. It is simply a pulse generator and its frequency is called the increment rate. The gated clock pulses drive a "low period" counter. A typical count period is 5 seconds. (This parameter is also available for adjustment by the operating engineer.) If this counter is not reset within five seconds of modulation time, the increment pulses will go to an up-down counter and cause it to count upwards. If either of these low modulation thresholds are satisfied within the five second period, the counter will be reset to zero and a new period will begin.

Thus, if the transmitter has not had at least one modulation burst exceeding 85% negative modulation or 100% positive modulation within 5 seconds of modulation time, the counter will increment and increase the program level driving the transmitter. This gain increase will continue until the low modulation level criteria have been satisfied.

Likewise, if the modulation exceeds minus 95% or plus 112% at any time, a decrement clock will be gated, causing the counter to count downwards reducing the audio level to the transmitter until the high modulation condition has been cleared.

The other two thresholds shown on the illustration measure the overmodulation conditions and are connected to two counters whose counts are displayed on the front panel. These counters operate for one minute so that the operator can see how many modulation bursts, both negative and positive, have occurred in the preceding one-minute period. They display both the accumulating count in the current minute and the total count in the preceding minute.

The counters are necessary and convenient tools for adjusting the variables for the desired degree of control. They also provide the necessary overmodulation count signals for ATS operation.

The actual program control is done by an eight-bit multiplying D to A converter. Without resorting to unnecessary detail, this device can be described as a linear attenuator, adjustable in approximately 0.1 dB steps by an eight bit binary word coming from the up-down counter. The total range of adjustment is plus or minus 8 dB. The adjustment steps are so small that gain changes are not discernible in the program material.

This then, although greatly simplified, is a complete description of the Amplitude Modulation Controller and its operation.

Field Test . . .

Figure 2 shows a prototype model of this device.

As you can see from the numerous knobs and dials on the front panel, we have made all of the parameters, threshold levels, the increment and decrement clock rates, the low modulation period counter time and the display counter periods adjustable for field evaluation.

The production models have all of these adjustments with the exception of the overmodulation counter periods adjustable behind a front panel door.

The range of adjustments have been restricted to practical values as determined in our field tests. By setting these adjustments in the desired manner, the unit can be made to operate very slowly as might be desired in a classical music station so that long term variations in transmitter characteristics can be corrected without any change in program material characteristics. It also may be adjusted for fast action so that very high average modulations can be obtained. This would be suitable for stations operating with more contemporary formats.

In the past several months we have made extensive field tests at three different broadcast stations.

Generally, we asked the station engineer to adjust his program equipment to obtain modulation characteristics according to his normal operating procedure and to include all of the audio processing equipment normally in the circuit. We then inserted the Amplitude Modulation Controller into the system and observed the adjustments that were made by the device.

The operating gain is displayed on a front panel meter. This is accomplished by running a small fixed dc voltage into the multiplying D to A converter and measuring the voltage delivered at its output. Thus, as the circuit adjusted itself for different gains, the through gain of the audio circuit is displayed on the front panel meter.

Figure 3 is a recording chart showing the voltage at the meter terminals over several hours of operating time. These records were made with the chart speed of 6 cm per hour and the real time is shown along the border of the chart. A scale calibrated in system gain is shown at the bottom of the chart.

The particular record shown was made at radio station WMAL here in Washington. The Amplitude Modulation Controller, as can be seen, increased the average signal to the transmitter by about 1 to 2 dB during most periods.

Several interesting features are clear on this record. In the area marked (1) it can be seen that as much as 3 dB of correction is made on changes of program source. That is, on a change of live to network, announcer to recorded music, etc.

Occasionally the gain is reduced to below zero to prevent overmodulation. A typical example of this is shown in area (2).

The large downward spike at (3) shows a malfunction of the equipment on pattern change. When the carrier momentarily went to zero, the logic system appeared to detect a very high modulation and drastically reduced the audio level. Subsequent to this experiment we installed a special circuit to idle the control logic when the carrier dropped below a predetermined level. This malfunction was not detected in further tests.

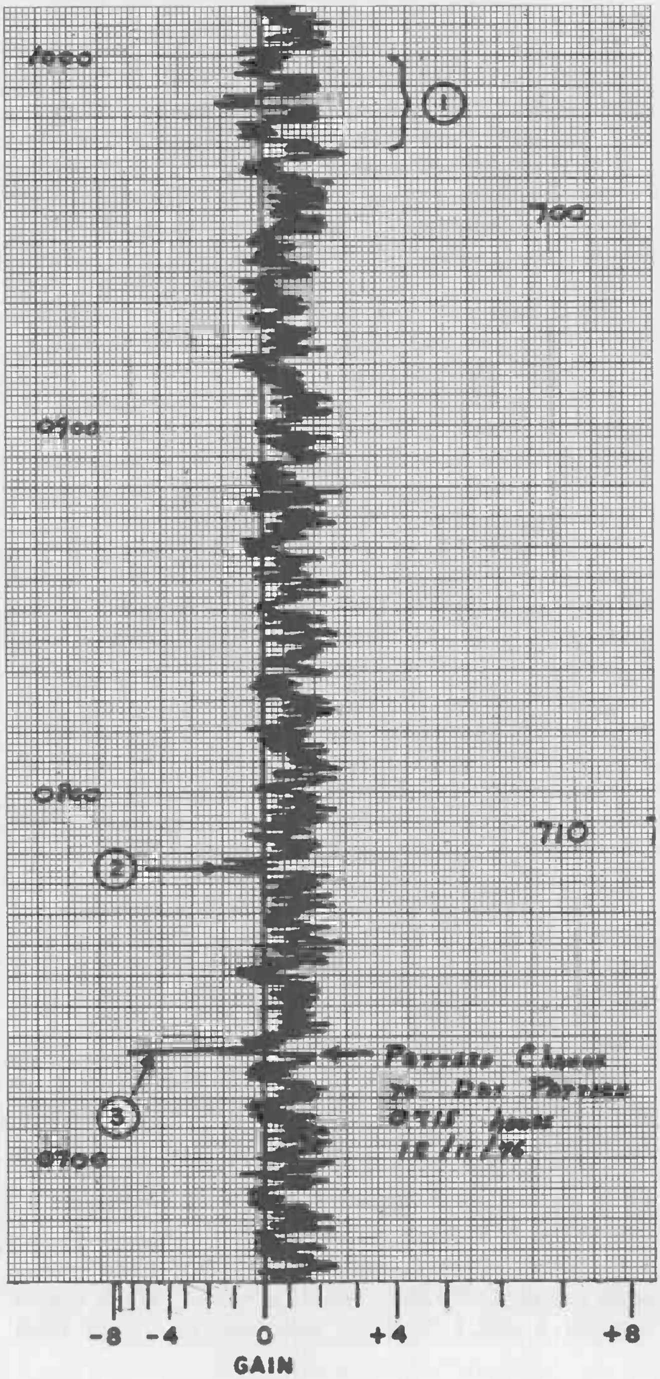

Figure 3.

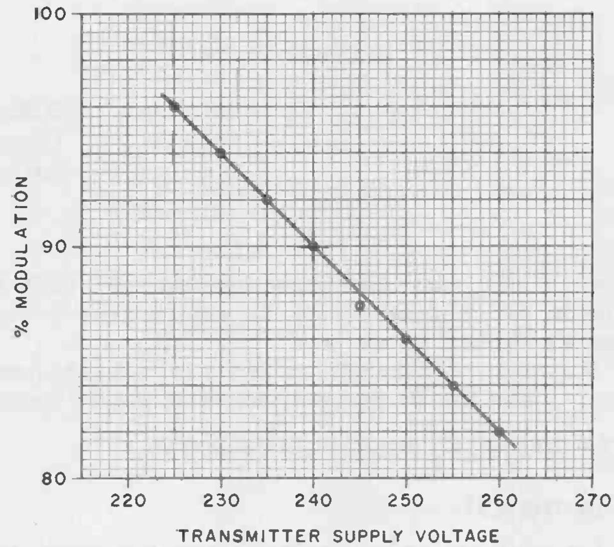

Figure 4.

It is of interest to point out that these corrections were made even though WMAL was operating with what it considered the best complement of audio processing equipment.

The tests at the other two stations showed results very similar to these. At one station the gain adjustment was larger because the audio and symmetry level varied considerably between program sources.

An interesting transmitter characteristic was measured at WGH, Hampton Roads, Va., during the tests. WGH is equipped with two complete transmitter sites and with diesel backup power. They were kind enough to permit us to operate one of the sites on a dummy load and vary the primary supply voltage to the transmitter using the diesel generator.

With an audio oscillator source we adjusted the transmitter for 90% modulation with a 1000 Hz tone. The audio level was maintained constant and the AC supply voltage was varied from 225 to 260 volts.

Figure 4 shows the measured percentage of modulation for different supply voltages. It is striking to see how the transmitter characteristics change under these conditions.

Mr. Looper, Chief Engineer for WGH, has long recognized this problem and in fact presented a paper at last year's convention on the subject. He has successfully corrected the problem by regulating the modulator bias and the dc supply to the audio amplifiers in the transmitter. It is obvious though, that standard audio processing equipment will not cope with this situation. In fact, Mr. Looper's paper on this subject provided the impetus for our development of the Amplitude Modulation Controller.

I would like to thank Mr. Looper, Mr. McPherson of WMAL Radio 63, and Mr. Miller of WFAX for their great interest in this project, for the use of their facilities for field testing, and for the suggestions and guidance they have given us in this program.

Care and Feeding of Directional Antennas

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Each and every directional antenna ever built is unique. There has never been nor will there ever be two that are identical in every way.

But all directional antennas have many things in common. These can be grouped into three general areas: *THEORETICAL, PRACTICAL and FCC.*

Theoretical . . .

Let me begin by saying there are no text books, no good "how to do it" books. Only experience can be the true teacher when it comes to the best way to learn. In fact, when I went through college I never had a single course in directional theory. But then Christopher Columbus never took a course in world geography, either.

Let's begin our theoretical study by defining a few terms.

The first is *Panic*. This is what sets in, in the mind of the chief, when he sees the consulting engineer driving away and he first realizes the whole thing is his responsibility.

Second is the term *Directional Antenna*. This I define as an antenna system consisting of more than one tower intended to restrict power in specific directions.

Let me define the term *FCC Rules*. This is a document created by people who suffer from Potomac Fever, and have no engineering logic in fact. Contrary to popular belief Volume III was not carried down from Mt. Sinai by Moses!

Next is the term *Pattern Polar Graph*. This I define as something which always works on paper, but can seldom be disproved in practice. Let me repeat that definition! **Something which always works on paper, but can seldom be disproved in practice.**

It is good for each chief to reach at least a basic understanding of just how his individual directional antenna is supposed to work.

In other words, the basic idea is useful and necessary in servicing and maintaining it. The real question is how do you do this?

This can be approached by each chief in several ways. *First,* by reading the original FCC application and its design data. *Second* would be to ask questions of your consultant. After all he should understand your system if he made it work. And *thirdly,* I believe the best way is to collect magazine articles and technical papers written about directional antennas.

Practical . . .

The second general area is, of course, the practical considerations of any directional antenna.

Obviously, the most important consideration is Preventative Maintenance. There is no substitute for good P.M.! In most cases it merely means doing the obvious.

For example, when the weeds and brush get so thick that reading the base current is akin to a jungle training course, it's time to clean up the antenna field. In other words, it should be obvious to keep the weeds, trees, bushes and climbing vines to a minimum.

I think it is also obvious that all snakes, mice and birds should be kept out of the dog house. Keeping connections bright, nuts and bolts tight, and movable parts lubricated should also be obvious.

When trouble occurs, don't panic, but inspect for physical damage. Many problems can be corrected or even forestalled by keeping your eyes open when you conduct the periodic physical inspections of your antenna system, required by the FCC.

For example, if you walk out to the tower and you observe bits of broken guy wire insulators lying in the grass it should ring a bell that something is wrong. In the case of KMPL we observed this condition and found some 70 insulators had been punctured by lightning.

Also if you lose the base current reading on tower Number 3, look to see if the copper tubing between the ATU and the tower has parted. This occurred at WGSB and the operator never did find it, until we were called.

But in some cases you don't have to be too sharp to find trouble. In fact, in one case I know, it came right out and hit the chief.

In the old days at WBBM, (because of tower height), it was not uncommon for lightning to strike the tower and weld the spark gaps. Standard procedure was to grab a small sledge, run out the cat-walk to the tower, and separate the gap.

One dark and stormy night the 50 kw went down and it could not be brought back up. So my friend grabbed the trusty sledge, pulled on his raincoat, opened the rear door and ran for the tower to separate the spark gaps.

Only halfway out something hit him in the chest. There was the tower laying across his path. This would be a case of obvious reason for trouble.

In the case of WJPS, they had a base current and phase angle problem with their Number 3 tower. While towers Number 2 and 1 exhibited some sort of change from licensed values, Number 3 reflected the biggest shift.

It is, of course, good engineering practice to physically observe the tower which changes the most, when one is trying to locate any fault. This is not always true, but most of the time it points to where the problem lies.

In this case the chief claimed he had tried everything. He checked parts, he looked for shorted spark gaps, base insulators, etc. He could find nothing after two weeks of searching.

When I arrived on the scene, it was obvious it had to be in Number 3 tower, since I observed base current flowing daytime when this tower is normally isolated. In walking around the tower base I quickly observed that the insulated phase sampling line had drooped to the point where it laid across the ATU metal cabinet. The RF had burned through the jacket of the RG 8/U cable, grounding out the tower. With one block of dry wood I was able to quickly return WJPS to licensed operation.

I trust these examples point up the fact that one should not panic, but carefully use all your God-given senses to look for trouble.

I recommend that once per year you check the accuracy of your base current meters.

One tower should be used as the reference tower. The other meters can be carried over to this tower and substituted one at a time, or a spare meter can be carried from tower to tower to serve as a comparator.

In addition to the advice of not panicking, I would say the second best advice is "fight temptations." Just don't start cranking the knobs on the phasor the minute some meter or monitor reads at variance. After all, there could be a logical reason why.

I recommend a wait and see approach. When all else fails, then tune the pattern, since there are so many, many things, like weather, that can result in temporary shifts in any array.

The case of a new engineer at WTAQ comes to mind. The first night he was on duty it was raining. And as any experienced operator would know, some of the readings shifted from normal. When he observed these changes he grabbed the cranks on the front of the phasor and attempted to bring things back. Of course, he failed, and the next day was seeking employment elsewhere. It only took the chief seven days of tuning to correct 15 minutes of haste.

With regard to your monitor points, don't be surprised if the normal signal intensities change when the characteristics of the vicinity shift.

An example is our 136° monitor point at WGSB. It used to be a wide open field with a good reading. One day Commonwealth Edison Company erected a nice 139 kv transmission line right through the monitor point. As you might expect the normal signal reading changed! In fact it changed by 300 percent upward!

Whatever the reason a meter changes, or a value shifts beyond normal range or limits, explain it on your log. **Write it down.** I must stress you can never be too "wordy" in such explanations. Here are some excuses I have commonly used. Blame it on the weather, on physical changes or, if all else escapes you, on the manager's dog!

FCC . . .

Let's talk about compliance with the FCC Rules, as they pertain to a directional system. Three basic areas need be observed: Base Current Ratios, Antenna Monitor Readings, and Field Monitor Points.

Base Current Ratios depend upon RF ammeter readings. When a meter reading changes from normal, its corresponding ratio will also change. Most stations have limits of $+5°$.

Let's see what the solutions are to base current problems. At one station which shall remain nameless, they had an unusual amount of wandering in the base current readings. This was solved, at WRSW, by a very innovative chief engineer. He merely purchased several extra meters and then labeled them (+) or (−) depending upon their accuracy. Now whenever a ratio gets too high or too low, he merely switches meters. It works great.

The second area of concern is with antenna monitors, more commonly referred to as "phase monitors." Here one need "hold" both phase angles and remote ratios.

Phase angles are usually set by the consultant at the time he originally tunes the pattern. But since none of us are perfect, they sometimes wander. This is easy to compensate for. Since any change obviously is due to aging or temperature variations in the length of the sampling lines, just correct for it. A box full of various phase lengths will easily bring the phase reading back to normal.

If you do not have the proper length of phase line to correct the error, you may employ this formula to put the meter reading right on the nose. Feet to add (or sub-

$$\text{tract}) = \frac{\text{Degrees} \times 2731 \times \text{Velocity}}{\text{Freq. (kHz)}}$$

With remote base current readings it is a little more difficult. If you have loops on the tower, they can be turned. Or if current transformers, they can be adjusted!

The last of the three areas is Field Monitor Points. With monitor points there isn't much you can do, except, of course if the readings are too high. In this case, try placing some weak batteries in the FI meter.

The last area I wish to discuss is what to do in case you can't keep the directional meters reading legal. This would logically necessitate a change in the license. If so, Form 302 is required.

When you fill this out **write legibly** and clearly. You can't write down too much information. Explain why things shifted or why they are not the same as called for by the license. After all everybody else has 20° phase shifts and 50 percent ratio changes too on sunny days.

In the event an FCC Inspector rejects your explanations be prepared to quote that famous motto of the hearing aid industry:

"Go stick it in your ear."

Now that I have shared my thoughts and ideas from 22 years of consulting experience, I leave you with this parting gem of wisdom,

"The college of hard knocks is still the best education — Unfortunately most of us can't afford the tuition."

New Techniques for Generation of Composite Stereo Signals

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and
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FM stereo radio broadcasting is rapidly becoming a highly competitive medium. This change manifests itself in many ways, including the attempt to have a loud but clean sound and a superior technical quality.

In this presentation we are introducing two truly new innovations which accomplish these objectives: the **Dynamic Transient Response** or DTR lowpass filter and the **Digitally Synthesized Modulator** or DSM stereo generator.

The DTR filter, due to its non-overshooting properties, allows a loudness increase of 2-6 dB with absolutely no degradation of audio quality.

The DSM stereo generator does not suffer from the primary limitations of conventional balanced modulator or switching type stereo generators—distortion products in the former and poor separation at high frequencies in the latter. Uncompromised separation, linearity, and spectral purity result from the new DSM technique.

The DTR Filter . . .

FM stereophonic broadcasting is a frequency domain multiplexed (FDM) system. A left-plus-right (L + R) signal is transmitted in the band 30 Hz-15 KHz. This is the monaural baseband signal. A double sideband suppressed carrier (DSB) signal modulated with left-minus-right information is transmitted at 38 KHz.

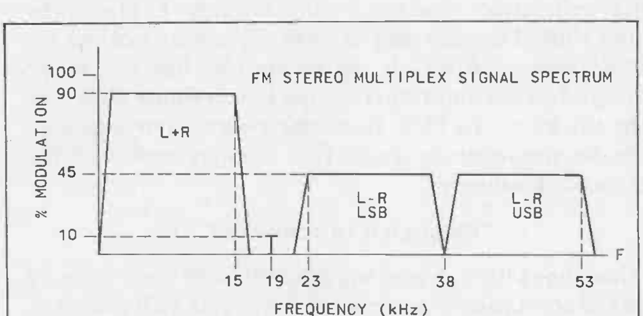


Figure 1.

To properly demodulate the DSB 38 KHz signal, a 19 KHz pilot tone is transmitted with a phase that, when it is frequency-doubled, L − R information can be synchronously detected. The composite stereo signal is shown in **Figure 1.**

Constraints are placed upon the amplitude and bandwidth characteristics of the left and right channel audio signals so the resultant L + R and L − R signals will exceed neither their amplitude bounds nor bandwidth allocations. Otherwise, the multiplexed signals would suffer distortion and mutual interference.

To control the amplitude of the L and R channel signals, AGC amplifiers, peak limiters, and clipping devices are customarily used. Typically these processors add to

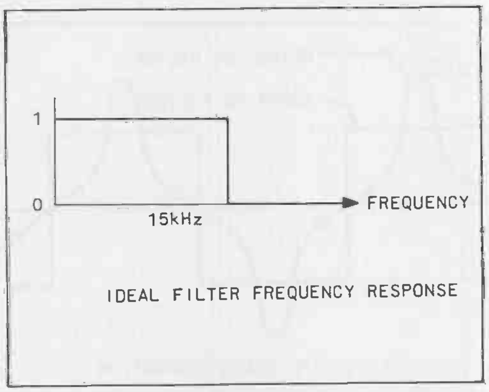


Figure 2.

the harmonic content of the program, producing a signal which would result in excessive bandwidth.

Any high frequency noise or clipping products from the limiter or program material beyond 15 KHz will be boosted by the preemphasis unless lowpass filtering is used. High frequency products interfere with (1) the pilot and (2) the L − R sidebands. The effect of pilot interference will have different effects on different receivers, varying from loss of stereo separation to complete break-up of the signal. Interference with the L − R sidebands will be perceived as noise in both channels correlated with the L or R signal.

The effect of this interference to stereo signals is termed "Dynamic Separation". Dynamic separation refers to the effects of aliasing in a frequency domain multiplexed (FDM) system.

A recording with a "hyped" high end frequency response after preemphasis may have signal components in the vicinity of 19 KHz at the same amplitude (or greater than) the pilot. (− 20 dB from 100% modulation). This will cause noise in older stereo demodulators and phase ambiguities and/or unlocking in newer PLL detectors. Similarly, the same recording may elicit limiting products near 38 KHz which will appear as a raspy noise in both channels.

If a particular commercial or record that is run often happens to cause these effects, difficulties may ensue. The advertiser may be anxious to know why his commercial sounds distorted in stereo. The listener may be upset that this weeks' #1 record is punctuated with bursts of correlated noise. The only universally effective way to solve the problem is through the use of audio lowpass filters to eliminate spurious products at 19 KHz and above.

In the better stereo generators, lowpass filters have been included to attenuate harmonics beyond the 15 KHz bandwidth of the system. Some inexpensive switching type stereo generators omit the audio lowpass filters in an attempt to eliminate overshoot. With no audio filters, the stereo composite lowpass filter will overshoot instead.

This is absurd. Not only has the overshoot problem been left unsolved, but the stereo generator is vulnerable to pilot interference and aliasing.

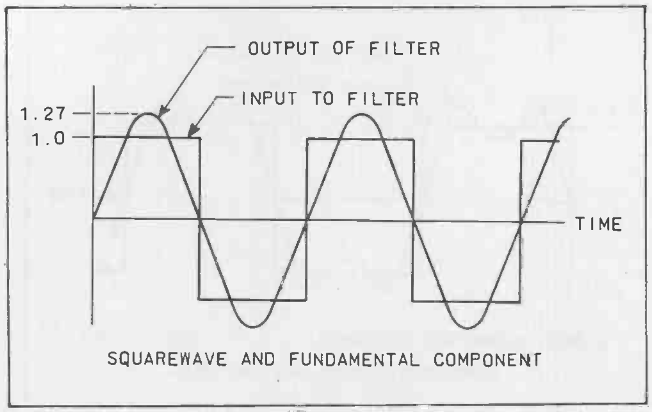


Figure 3.

Causes and Effects of Overshoot . . .

Although the input to a lowpass filter may be accurately amplitude-limited, this is not necessarily the case at the filter's output. Ringing and overshoot of the filter can seriously degrade the accuracy of the limiting action. Lowpass filters may overshoot 6 dB (100%) on some signals which are not uncommon at the output of audio processing equipment.

A lowpass filter changes two independent qualities of its input signal. In addition to the obvious change of the amplitude vs. frequency characteristic the filter also changes phase relationships among different frequencies in the filter's passband. This is equivalent to stating that different frequencies take different lengths of time to propagate through the filter. Associated with these two changes to the signal are two mechanisms causing overshoot.

Attenuation of Harmonics . . .

Consider the ideal case of a lowpass filter with rectangular frequency response and zero time delay. This filter is in fact unrealizable, but nevertheless would exhibit overshoot due to elimination of harmonics. The frequency response of this filter is shown in **Figure 2.**

Assume that the input signal is a 10 KHz squarewave of amplitude A. The Fourier expansion of this signal is:

$$v(t) = A \frac{4}{\pi} \sum_{n = 1, 3, 5, \ldots}^{\infty} \frac{1}{n} \sin(2\pi fnt)$$

where f is frequency. The squarewave signal has components at the fundamental and odd harmonic frequencies only, i.e., 10, 30, 50, 70, etc. KHz. Since the filter cuts off at 15 KHz only the fundamental (10 KHz) component of the squarewave appears at the filter output.

Note that if the squarewave amplitude (A) is one volt, then the peak value of the fundamental component (identically equal to the output signal) is 4/pi or 1.273. This constitutes an overshoot of 27%.

The squarewave and its fundamental component are shown superposed in **Figure 3.** This is only one example of many possible signals that would cause a linear phase lowpass filter to overshoot.

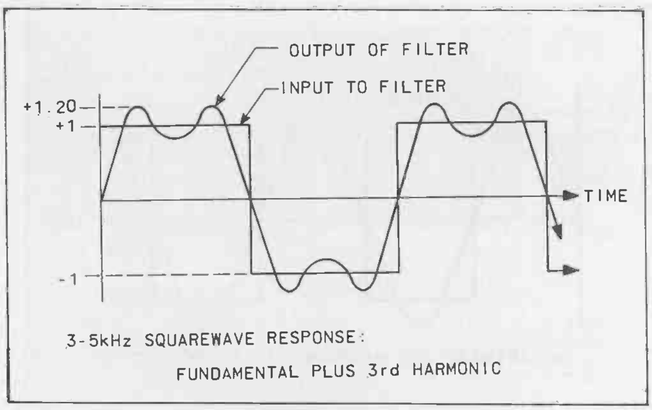


Figure 4.

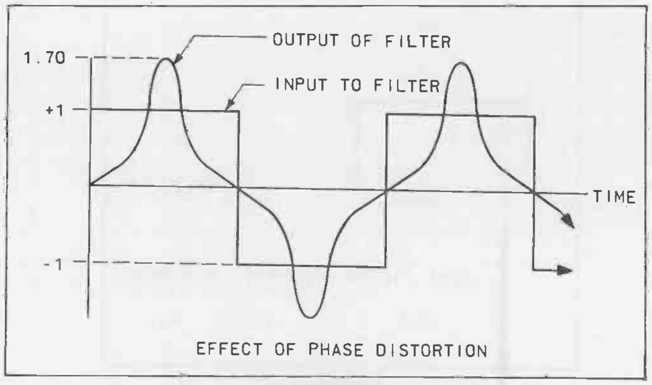


Figure 5.

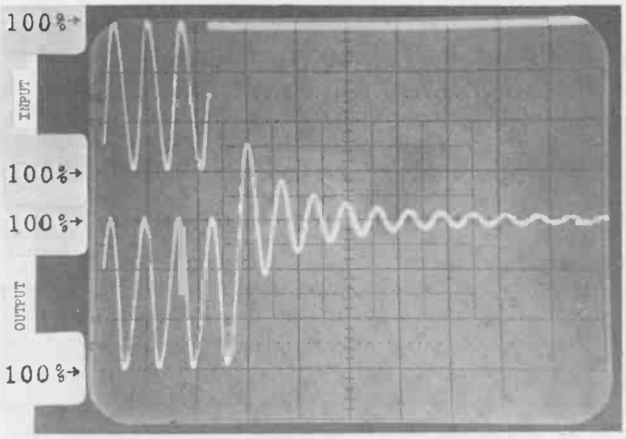


Figure 6.

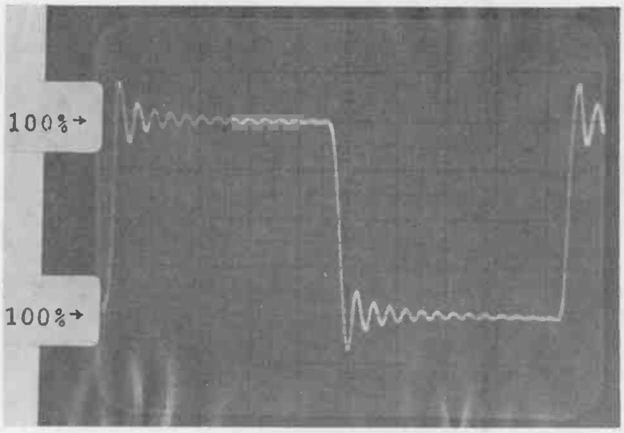


Figure 7.

Non-Uniform Time Delay . . .

If different signals propagate through the filter with different time delays, it is possible for input signals separated in time to become coincident at the filter's output. This could result in an overshoot.

Continuing with the example of the squarewave, consider the case where only the fundamental and third harmonic fall within the filter's passband. Squarewaves in the range of 3-5 KHz satisfy this condition.

The input and output of the ideal filter are shown in **Figure 4.** Overshoot is 20.0%. Since such a filter is impossible to build, the response of Figure 4 in general cannot be produced. Rather, time delay will vary as a function of frequency, thereby upsetting the phase relationship between the fundamental and third harmonic.

If the phase of the third harmonic is shifted 180 degrees, the waveform in **Figure 5** results. Overshoot is 70% or 4.6 dB.

Time delay generally increases with frequency within the passband of an elliptic filter. Minimum time delay is at DC while maximum time delay within the passband occurs at the cutoff frequency.

A test signal has been devised which causes filters to overshoot primarily as a function of their time delay distortion. The test signal consists of a sinewave burst immediately followed by a DC step signal. The sinewave will accumulate maximum time delay (2.38 usec) while the DC step signal will accomulate a minimal time delay (43 usec.) At the filter's output the sinewave will coincide with the beginning of the DC step signal.

This phenomenon is shown in **Figure 6.** Note that the overshoot is 100% (6 dB)!

Through a combination of effects (both attenuation of harmonics and nonuniform time delay) a myriad of signal types can cause a typical elliptic lowpass filter to overshoot.

A low frequency squarewave response is shown in **Figure 7.**

The waveforms of Figures 6 and 7 are common with certain types of music and/or certain types of FM limiters. To offset overshoot, audio levels are simply turned down to a point where overshoots "of frequent recurrence" do not exceed 100% modulation. This can mean a sizeable reduction in modulation effectiveness, usually on the order of 2½-6 dB!

Need for a New Approach . . .

There have been several previous approaches to the problem. Although existing systems do control overshoot, they also contribute unwanted side-effects to the signal.

One method for overshoot control uses a delay and an AGC stage. This system can cause gain "pumping".

Another very popular system uses alternate clipping and filtering combined with a complementary high frequency boost and cut. This system suffers from excessive intermodulation distortion and a high frequency rolloff which is dependent upon signal level.

It is clearly desirable to have a filter which will eliminate harmonics above 15 KHz yet preserve the peak amplitude-limited nature of its input signal.

Note that it is *not* necessary to have a filter that does not overshoot. Ringing and overshoot are completely unobjectionable provided the overshoots do not exceed the 100% modulation level.

From this point on, the term "overshoot" will denote only overshoots above 100% modulation. The filter requirements are:

- Frequency response flat ± ½ dB 20 Hz-15 KHz at all levels up to 100% modulation.
- Attenuation above 19 KHz inclusive: 50 dB minimum.
- Overshoots not exceeding 102% modulation.
- Filter shall be transparent to steady state sinewave signals. THD and IM distortion 0.1% or less.
- Any effect of eliminating overmodulating overshoots shall be inaudible.

Until now, there has been no filter or system which accomplishes all of the above objectives. Deficiencies in each of the existing overshoot control schemes necessitated the development of a new filtering technique. The DTR filter is the only method which satisfies all of the above criteria.

Performance . . .

The DTR filter compensator is highly effective in its operation.

Overmodulation due to overshoot has been reduced to 2%, approximately the accuracy of a modulation monitor. Operation is possible with *any* limiter and any program material including Dolby® processed audio.

Setup is easy. All one does is apply a signal that is known not to overshoot at 100% peak modulation. This can be a 400 Hz sinewave. Using this signal as a reference the compensation thresholds are set to a level corresponding to 100% modulation. LED indicators are provided to indicate when an overshoot is compensated, thereby facilitating setup.

Extensive listening and A-B tests have shown that on all types of programming the DTR filter produces no audible effect.

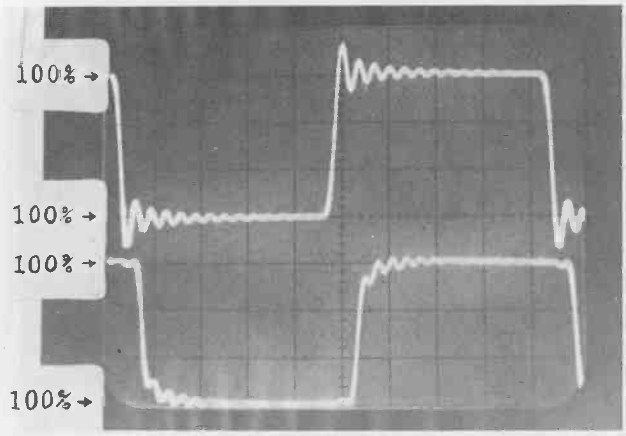


Figure 8.

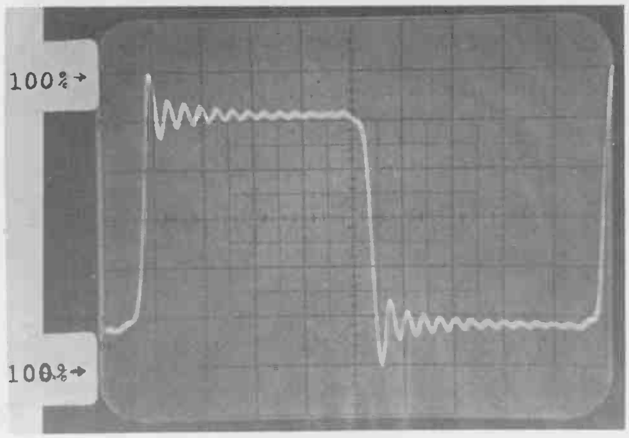


Figure 9.

Unlike other techniques using AGC, delay lines, or conventional clippers, the DTR filter takes action if and only if an overshoot is imminent. Because the energy contained in the overshoots is inappreciable, deletion of that energy is imperceptible to the ear.

The modulation monitor, however, does not respond to average energy but rather peak voltage. The perception of the modulation monitor (and the FCC) is unlike that of the ear and *does* respond to low energy, high instantaneous amplitude transients.

The DTR filter overshoot compensator function removes the transients above 100% modulation. It does no more and no less.

When a 600 Hz squarewave is applied to a conventional 15 KHz elliptic function lowpass filter, the output rings and overshoots as shown by the top trace of **Figure 8**.

The DTR filter is shown in the bottom trace. There is a small amount of ringing; however, none of this causes overmodulation.

Figure 9 shows the same squarewave but applied at 65% modulation. The system is completely linear at this level and no compensation is taking place.

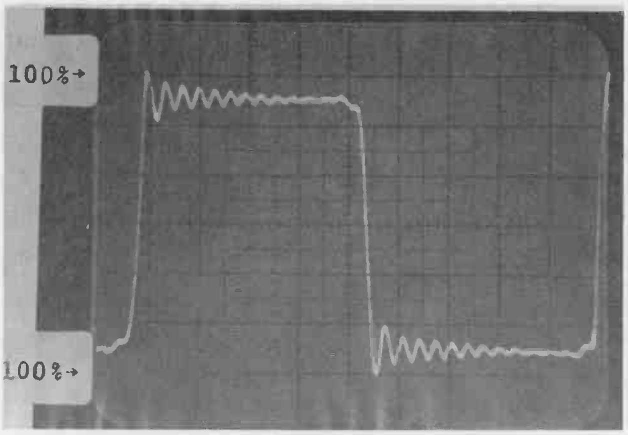


Figure 10.

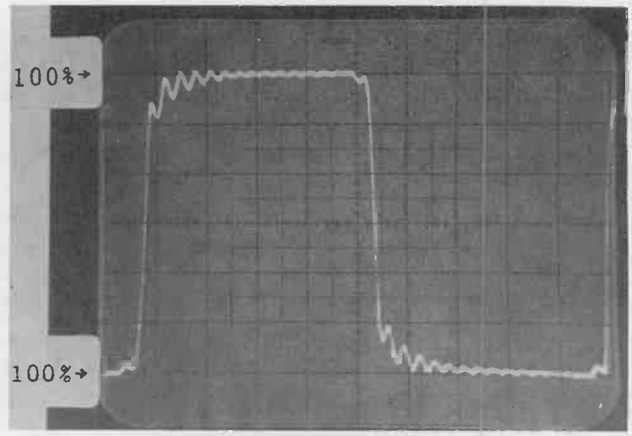
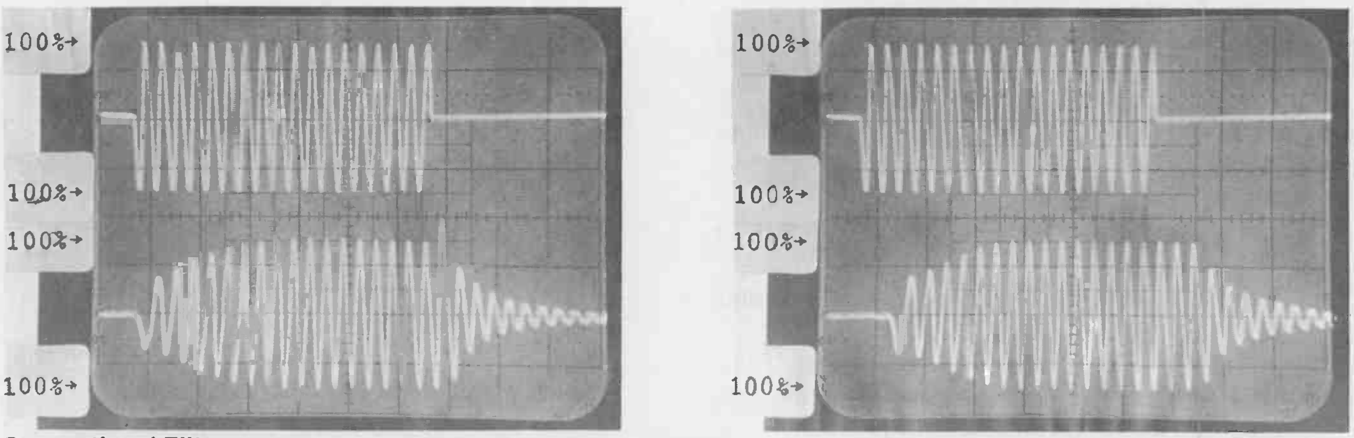
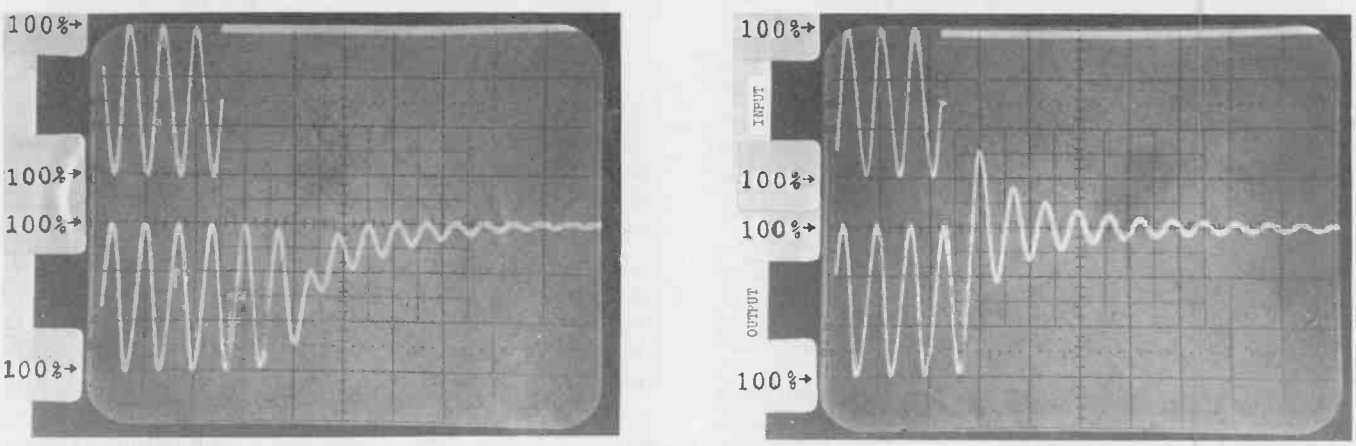


Figure 11.



Conventional Filter **Figure 12.** **DTR Filter**



Conventional Filter **Figure 13.** **DTR Filter**

In **Figure 10** the level has been increased to 90% modulation. Here there is some action taking place to limit the first cycle of ringing to 100%.

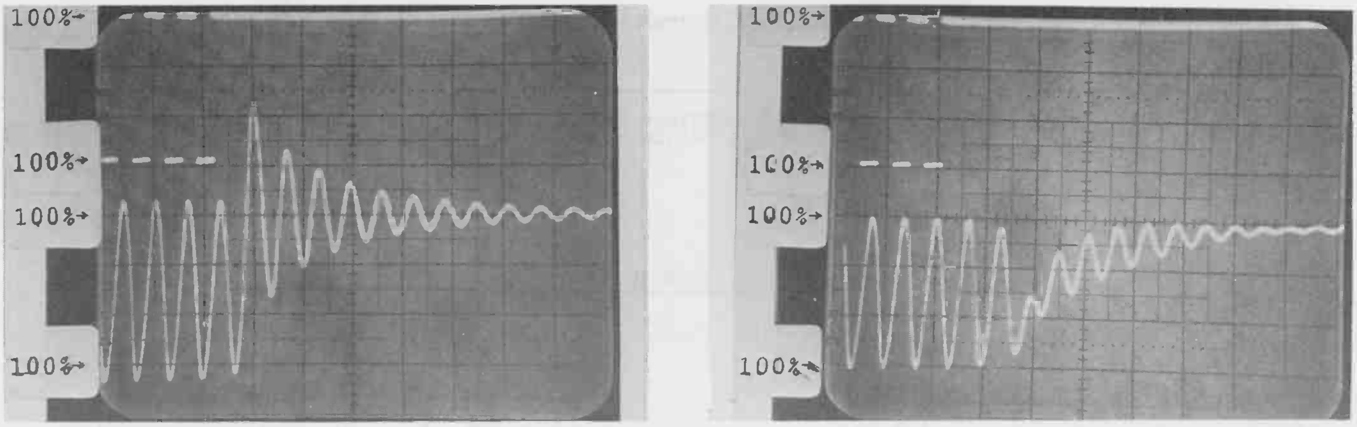
In **Figure 11** the overshoot compensator is completely enabled as the squarewave is applied at 100% modulation.

Tone bursts also demonstrate the capabilities of the DTR filter. **Figure 12** shows at left a 15.0 KHz tone burst input signal and the output of a conventional filter. The output of the DTR filter is shown at right.

One of the most difficult tests to be contrived for a filter is the sine/step signal described earlier.

Figure 13 shows at left the input and output of a conventional filter. Overshoot is 100%. If the modulation level of the transmitter were to be tuned down to accommodate such overshoots without causing overmodulation, a full 6 dB of signal would be lost!

The DTR filter response to the same test signal is shown at the right. Overshoot is less than 2%. Notice that the DTR filter has no effect on the sinewave.



Conventional Filter **Figure 14.** **DTR Filter**

It is possible to cause even more overshoot by substituting a squarewave for the sinewave part of the test signal.

Figure 14 shows at left the resultant squarewave/step test signal applied to a conventional filter which overshoots 140% or 8 dB! The response of the DTR filter to the same test signal is shown at right.

This kind of signal is not uncommon with many types of limiters. Bass note attacks simultaneous with sibilance have been known to cause 100% overshoots when processed by a clipping-type limiter.

Figure 15 shows the output spectrum of the DTR filter under near "worst-case" programming conditions.

The input signal had extreme high frequency content, and was processed by several limiters, the last being a hard clipper. The spectrum analyzer, a Tektronix 7L5, is equipped with a microprocessor and memory which was used to produce a maximum-hold display.

Over a three minute period the output spectrum of the DTR filter was continuously monitored; each time that a frequency contained more energy than previously, the display would be updated. The display of Figure 15, therefore, does not represent a typical instantaneous spectrum, but rather the maximum spectral amplitude over a three-minute period.

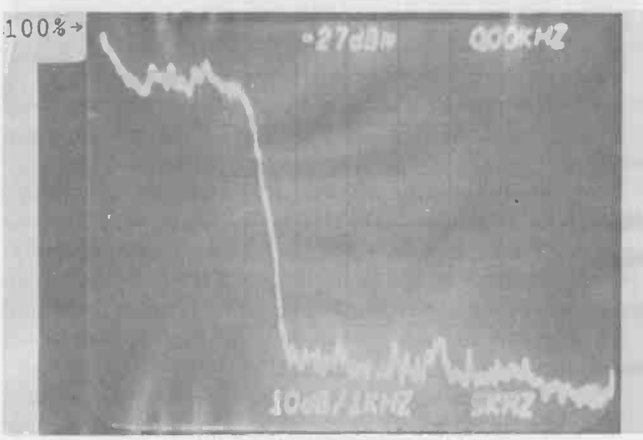


Figure 15.

Conclusion . . .

It has been shown that the DTR filter is a universally effective method for eliminating overmodulation due to overshoot. Because the compensator takes action *only* when necessary to remove low-energy overshoots, there is no audible distortion under any conditions. Because the DTR filter stands alone, a "systems approach" is not required; any FM limiter may be used.

A typical procedure for setting FM modulation levels has been to apply a sinewave through the program limiter at 100% modulation. A sinewave signal will not cause overshoot. Upon application of programming, the exciter filters will overshoot. With programming, the modulation level must be reduced 2½ to 6 dB to ensure that "peaks of frequent recurrence" do not cause overmodulation. With the Harris MS-15 exciter this last step of turning down the modulation has been eliminated.

The amount of loudness increase is a function of limiter type. The amount of overshoot that a limiter causes in a conventional filter is the amount of overshoot that is eliminated by the DTR filter. This same amount of overshoot is the increase in loudness that can be obtained. Some limiters, due to their excessive rolloff of highs, may only cause overshoots of 35% (2.6 dB). Other limiters, which rely more exclusively on clipping for pre-emphasis protection, can cause 100% overshoots (6 dB). Processors which rely upon a combination of clipping and rolloff will benefit from an intermediate loudness increase. Again, this causes no loss of audio quality.

Just as a conventional audio lowpass filter eliminates high frequency distortion products from the programming, the DTR filter establishes spectral purity. Signal quality commensurate with the DTR filter requires the new level of performance offered by the new digitally synthesized stereo generator.

The DSM Stereo Generator

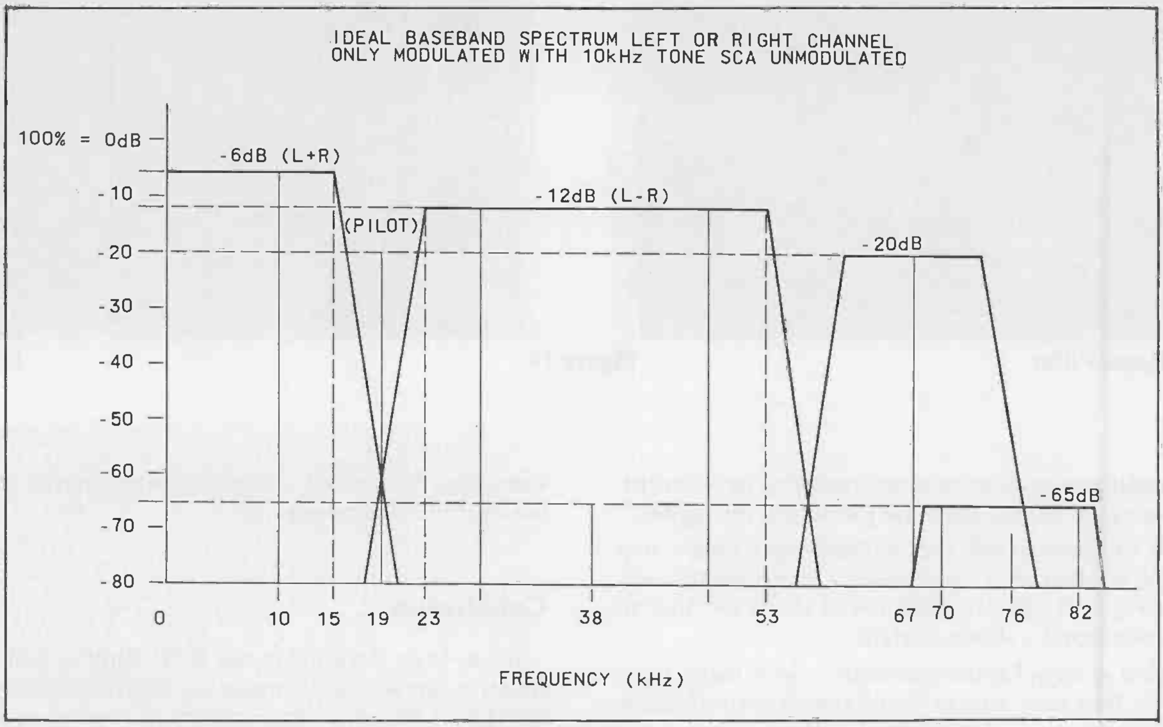


Figure 16.

Classically, there have been two methods to generate the composite stereo signal.

One method uses a system of matrices to develop the (L + R) and (L − R) audio signals. The matrixed (L − R) signal is coupled into a balanced modulator which generates the 38 KHz double sideband (L − R) signal. The output of the balanced modulator is summed with the (L + R) signal and pilot to produce the composite baseband.

The balanced modulator function can be performed by the familiar diode ring modulator, or more recently by the Gilbert transconductance multiplier. The key limitations to the performance of either of these systems is imperfect linearity and maintenance of the critical system balance with temperature and component aging. Nonlinearity of the balanced modulator causes both distortion to the (L − R) sidebands and generation of harmonic sideband frequencies which cause interference to the SCA, as well as an undesired increase in occupied bandwidth.

Figure 16 illustrates the ideal composite baseband signal with the addition of spurious products. Note that the lower sideband of the second order term about 76 KHz will fall directly on top of the SCA carrier when either stereo channel is modulated by a 9 KHz component.

Figure 17 shows the spectrum analysis of a recently introduced stereo generator using the Gilbert Multiplier balanced modulator. A considerable amount of spurious 76 KHz content is visible. Drift in the balance of the

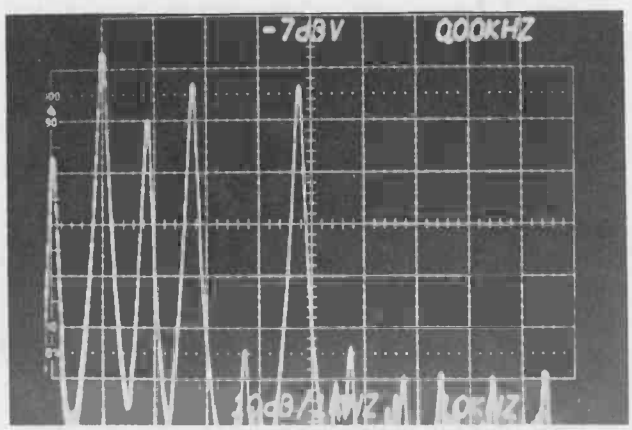


Figure 17.

modulator causes a decrease in 38 KHz carrier suppression and a shift in the (L + R) to (L − R) amplitude ratio which causes a decrease in stereo separation.

These balanced modulator stereo systems also suffer from a host of interacting adjustments as well as poor rejection of the harmonics of the 38 KHz modulating frequency. Furthermore, it is doubtful that the balanced modulator type stereo generators would permit optimal generation of discrete quadraphonic baseband without the addition of complex filtering.

The other classical method of FM stereo generation is the switching type of modulator. Here, an electronic switch alternately samples the left and right channel signals at a 38 KHz rate. The output of this switch must be lowpass filtered to remove harmonics of the switching frequency.

Figure 18 shows the spectrum analysis of a typical switching type stereo generator. Note once again, the spurious 76 KHz components due to lack of switching symmetry.

Even if the switching were perfectly symmetrical, thus producing no spurious components at 76 KHz or any other even order terms, there will be a third order term that produces a set of sidebands at 114 KHz which are only about 9.5 dB below the desired sidebands about 38 KHz. Since a 15 KHz modulation component in either channel would produce a component at 53 KHz (L − R, USB) plus another spurious component at 99 KHz (3X 38-15 KHz 3rd order LSB) due to the squarewave sampling function, the lowpass filter required to limit the spectrum to the desired components must pass 53 KHz with very little attenuation and simultaneously reject 99 KHz with very high attenuation. Furthermore, the lowpass filter must have a constant amplitude phase linear characteristic from 30 Hz to 53 KHz.

Filters of this type used in systems that conform to FCC specifications, are not only complex and expensive, but also degrade high frequency separation due to deviations from the necessary linear phase response.

The DSM Approach . . .

The DSM stereo generator reduces the balanced modulator function to a digital switching process.

A new sampling function shown in **Figure 19** has been developed which is much simpler to filter than the squarewave chopped signal. This new sampling function is a third order digital approximation to a sinewave. After only a moderate amount of filtering, the exact stereo baseband is obtained.

A spectrum analysis of the composite baseband produced by the DSM stereo generator is shown in **Figure 20**.

The four-step approximation to sinewave multiplication does *not* produce the 114 KHz by-product of the switching process. Therefore, while the switching type stereo generator filter must have flat amplitude and linear phase to 53 KHz while rejecting 99 KHz, the DSM filter must also have flat amplitude and linear phase characteristics to 53 KHz, but need only reject 175 KHz and above.

The filter transition band has been increased from 53-99 KHz to 53-175 KHz, a factor of 2.65. Filtering has therefore been made considerably easier.

With filter complexity and economics no longer limiting stereo performance, compromises need no longer be made in the overall performance of the stereo generation system. Stereo separation remains uniform over the entire audio range.

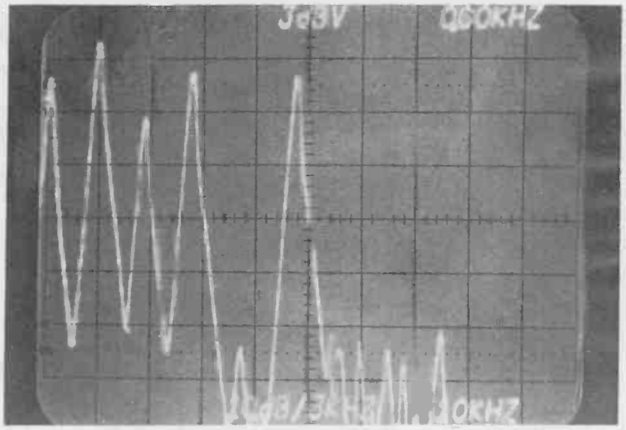


Figure 18.

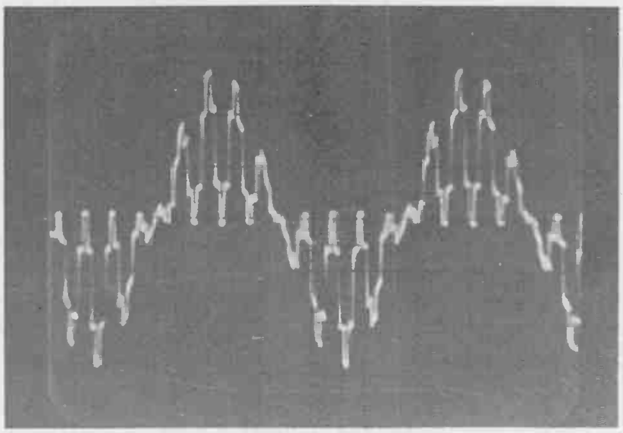


Figure 19.

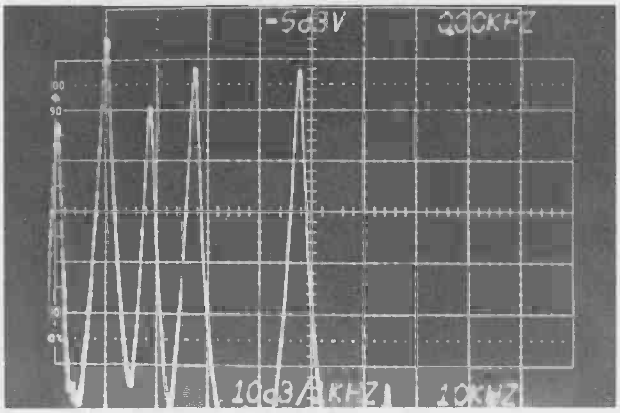


Figure 20.

Further analysis also shows that the four step sampling function is nine times less sensitive to separation adjustments than is the switching type of stereo modulator. The broadcaster can expect a stereo separation adjustment that is nine times less critical and nine times more stable than in previous types of stereo generators.

Due to the inherent stability of the DSM generator, even deliberate misadjustment of the separation control will not degrade stereo separation to less than 35 dB.

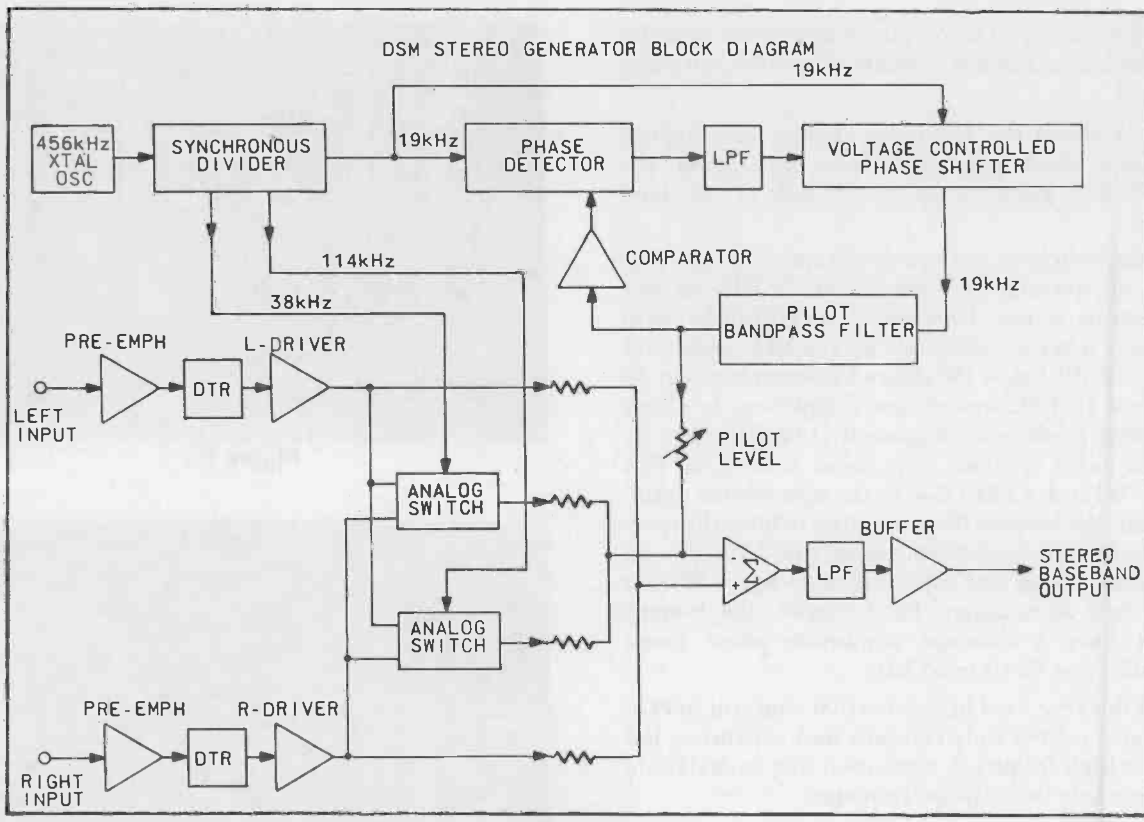


Figure 21.

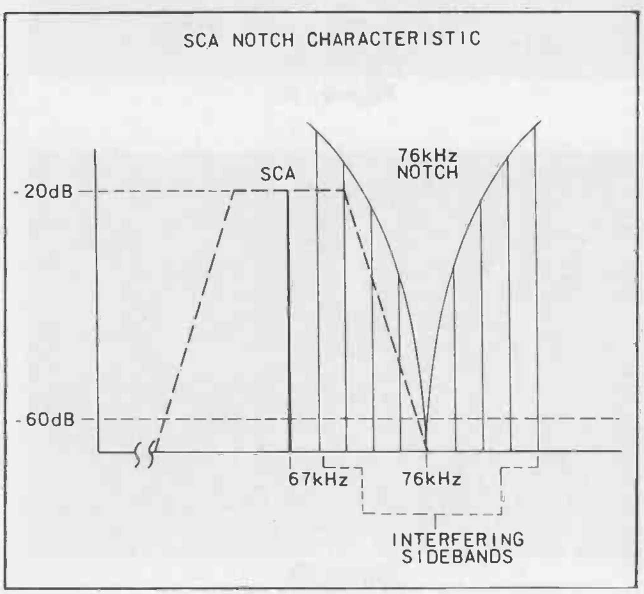


Figure 22.

A block diagram of the DSM stereo generator is shown in **Figure 21**. The DSM stereo generator contains a synchronous digital divider providing 19, 38, and 114 KHz outputs, electronic switches, a summing amplifier, and a phase linear lowpass filter exhibiting a uniform amplitude response and phase delay to 53 KHz.

A submultiple of the 456 KHz crystal oscillator provides the time base for the digital divider, while the synchronous nature of the digital divider ensures that all transitions from high to low and low to high occur precisely at the same time, thus assuring correct phase relationships among the three frequencies.

In addition, the configuration of the synchronous digital divider assures perfect symmetry of the sampling waveforms and provides excellent suppression of all even order terms including the 76 KHz sidebands which cause interference to the SCA.

Figure 22 illustrates the results obtained with non-DSM stereo generators using a 76 KHz notch filter to attenuate interference to the SCA. Note that while the 76 KHz carrier is inside the attenuation notch, the upper and lower second order sidebands of the (L − R) component, fall outside the maximum attenuation region of the notch. A 76 KHz notch filter will also disturb the amplitude and phase linearity of the stereo signal.

Referring back to Figure 20, it can be seen that the rejection of second order 76 KHz carrier and related sidebands, is constant and independent of the modulating frequency.

Figure 23 illustrates a spectrum analysis of the entire baseband including both the DSM stereo baseband and a modulated 67 KHz SCA. The guard bands between the (L + R) and the L − R) signals as well as the protection between the (L − R) and SCA signals are clearly evident. Unlike other types of stereo generators which utilize a 76 KHz notch filter, the DSM stereo generator provides excellent suppression of the 76 KHz sidebands independent of the modulating frequencies applied.

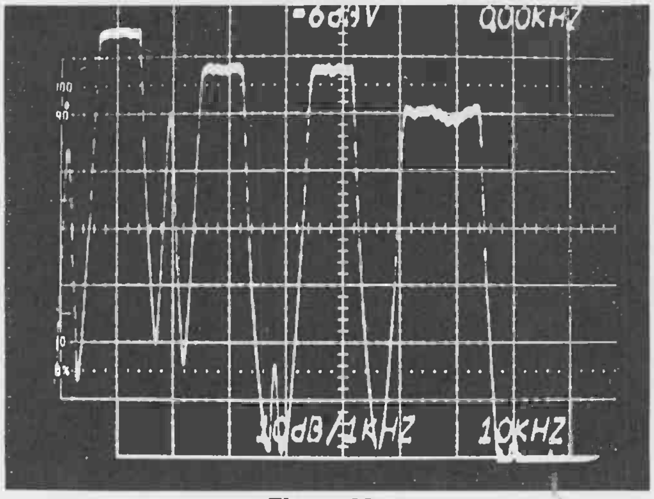


Figure 23.

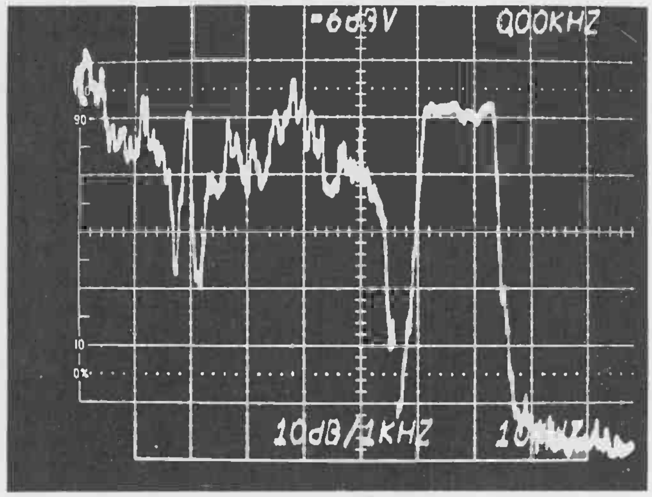


Figure 24.

Therefore, a reduction in stereo to SCA crosstalk is realized by the use of a DSM stereo generator.

The capability of any stereo generator to maintain high stereo separation is also dependent on its ability to maintain exact coincidence of the 19 and 38 KHz zero crossings. Due to the degree of accuracy necessary in maintaining this exact phase relationship and the problems associated with component drift with temperature variations and component aging, an automatic control system is necessary for long term phase stability.

The DSM stereo generator utilizes a phase locked loop which controls pilot phase independently of pilot filter tuning or drift in other components associated with pilot tone generation. The block diagram (Figure 21) shows this phase locked loop is rather unconventional; it employs a voltage-controlled phase shifter rather than a voltage-controlled oscillator. Automatic and precise control of pilot phasing combined with the highly stable (L + R) to (L − R) gain ratio of the DSM stereo generator in excess of 50 dB.

When the digitally Synthesized Modulator is combined with the Dynamic Transient Response filter (DTR), a new higher level of performance is achieved in several other areas.

High dynamic stereo separation is obtained. Dynamic separation is a measure of stereo separation under actual programming conditions often with highly processed audio. Unlike separation measurements made with sinusoidal signals, dynamic separation depends not only on the pilot phasing and correct (L + R) to (L − R) amplitude ratios, but also on the stereo generator's ability to protect the 19 KHz pilot tone as well as the (L − R) channel. Spurious information extended by the (L + R) channel degrades performance.

Figure 24 demonstrates how the DTR filter system attenuates super audible terms by more than 60 dB at both the pilot frequency and throughout the 23 to 53 KHz region. This results in dynamic stereo separation in excess of 40 dB under all normal programming conditions.

The DTR filter also limits the upper sidebands of the (L − R) subchannel to 53 KHz, thus preventing the upper sidebands of the stereo subchannel from extending into the SCA region. This protection against sideband interference combined with the excellent 76 KHz suppression inherent to the DSM system, provides stereo into SCA crosstalk rejection approaching 60 dB.

Figure 25 highlights the performance characteristics of the combined DTR and DSM systems.

PERFORMANCE SUMMARY

DYNAMIC TRANSIENT RESPONSE FILTER (DTR)

1 ABSOLUTE OVERSHOOT CONTROL (2%MAX)
2 NO AUDIBLE EFFECT
3 WORKS WITH ANY FM LIMITER
4 NO HIGH FREQUENCY ROLL OFF
5 PROVIDES HIGH DYNAMIC SEPARATION (>40dB TYPICAL)
6 OVERSHOOT COMPENSATION ONLY WHEN NECESSARY

DIGITALLY SYNTHESIZED MODULATOR (DSM)

1 EXCEPTIONALLY CLEAN BASEBAND
2 HIGH SEPARATION OVER ENTIRE AUDIO RANGE (>50dB TYPICAL)
3 AUTOMATIC PILOT PHASE CONTROL
4 HIGH STABILITY
5 SUPERIOR SCA COMPATABILITY
6 TECHNIQUE EXPANDABLE TO QUAD

Figure 25.

Quadraphonic Transmissions . . .

Because of the extremely clean baseband provided by the DSM system of stereo generation and the stable digitally synthesized nature of the system, compatibility with any of the proposed discrete quadraphonic transmission systems is assured.

The Harris MS-15 FM Exciter mainframe is already wired for plug-in quadraphonic modules, making the transition from stereophonic to quadraphonic transmission simple for the broadcaster.

Microcomputers in Television Broadcast Automation

Mitch Derick
The Grass Valley Group, Inc.
Grass Valley, Calif.

The advent of the microcomputer has made possible a revolution in television station automation.

The low cost and high reliability of the microprocessor chip makes it possible for today's automation system to be designed and built in a manner that was economically impossible a few years ago.

Computerized television station automation systems have been around for over 15 years, but the function that the automation has had to perform has remained the same. (i.e., Maintain the on-air continuity of a station's broadcast by automatically monitoring and controlling the master control switcher and its associated sources.)

Then and Now . . .

Yesterday's automation systems performed this task by using a single computer to monitor and control all functions. This was far from ideal.

While a few stations used large computers, most used minicomputers.

The large computer system, while being vastly more reliable than the mini (because of internal diagnostics), was so expensive that many other services had to be simultaneously performed to cost justify the equipment—accounting, inventory, billing, etc. At any moment any one of these other programs could fail and take the automation function with it.

The lower cost minicomputer, on the other hand, was dedicated completely to running the station automation system. Every function performed by the automation system was handled by a single semi-expensive minicomputer. If that one computer had problems, your automation was off the air. This, too, was far from ideal—but it did do the job.

Today's microcomputers are just as "powerful" as minicomputers. You could therefore replace the single processor "mini" with a single processor "micro." But what would be the advantage?

A few thousand dollars would be saved in the cost difference of the two processors, but that would be just about all. You would still have a single processor system.

What, then, is the ideal television station automation structure?

A Network Approach . . .

A *"Distributed Network"*—that is, "multiple computers connected together as a single system with each computer performing a specific task."

This concept is not new, nor is it limited strictly to microprocessors.

In the data processing world it is common practice to connect multiple "minis" and "maxis" into large, powerful, distributed networks. It is also common practice to spend multiple millions of dollars on such networks.

As a matter of fact, many television stations have daily contact with this type of distributed network through traffic systems. Subscribing stations usually have a "mini" on site which is connected to a "maxi" in another part of the country.

The greatest single advantage of the microprocessor, then, is to allow an automation system to be configured into a low-cost distributed network. There are two primary benefits which manifest themselves.

Like Building Blocks . . .

The first benefit is the "building block" concept. A distributed network is by its very nature a "building block" system. Since each processor performs a specific task, it can be separated into one stand-alone building block.

The advantage over yesterday's automation systems is immediately apparent. Instead of a single minicomputer performing all of the automation functions, multiple microcomputer building blocks can now do the same overall job.

Just as a child playing with blocks can build a simple structure and increase its complexity at will by adding more blocks, so can a station now move into automation gradually. Instead of acquiring a complete automation system all at once, a station could obtain only the building block which would allow the master control switcher to have "intelligent preroll."

Later on, the addition of another building block could allow pre-programming of a single break.

More building blocks would allow: a whole day or week's worth of programming, traffic system interface, material identification, log printing, etc.

The small station can acquire few blocks, the large station many blocks.

The analogy of a child playing with blocks can be carried even further. No two stations have identical requirements but, just as an almost infinite number of shapes and structures can be built out of a limited number of building blocks, so can many varying and different automation needs be satisfied with various arrangements of standard automation building blocks.

What you basically have is a "customized" system without paying the high price of custom development. The threat of obsolescence is also minimized.

Most television equipment becomes obsolete not because it will no longer function but because it does not have the latest, absolutely necessary "horn or whistle." With the distributed network building block approach, a new "horn or whistle building block" is simply added to the existing system by adding a new block or maybe simply modifying the "firmware" of an existing block.

Error Detection . . .

The other primary benefit is that of fault finding.

The single processor automation system has its hands full. Since the processor must run every facet of the system, it may be very busy during the most critical times of operation, with little or no processing time left over for error detection during critical moments (prerolls, transitions, etc.).

There is also the problem of limited memory capacity. A minicomputer that does everything must keep a large segment of its program, plus events, in its memory. This leaves very little room for diagnostic software.

The distributed network building blocks do not have this problem.

Each processor plays a single part of the whole operation and therefore has the time and memory to check itself, even during critical periods.

Note that the emphasis is placed upon error detection. When errors are detected, a human operator can be informed of the nature of the error and what steps to take to immediately perserve the continuity of the outgoing broadcast.

The ability of the system to notify a human operator of trouble is the tip of the iceberg of a distributed network's most powerful fault-detecting tool since it infers that the separate building blocks are aware of each other's problems.

This concept is made readily apparent by asking the question, "What happens to yesterday's automation system if the single minicomputer running it "dies" all of a sudden? Nothing.

The entire system could die two second before a $10,000 commercial is to air. All intelligence is gone. There will be no error messages.

In a distributed network automation system, each processor not only examines itself to make sure it is working correctly; but it also asks all the other processors in the system how they are doing.

If a building block partially fails, it will notify the operator of its problems. If it completely fails, the other building blocks will notify the operator of its "death."

The addition of more building blocks does not weaken, but strengthens the automation system.

Conclusion . . .

There are, of course, many other advantages of a distributed network microcomputer building block automation systems but the above two benefits are the most apparent.

The microcomputer in television electronics is here now and is here to stay.

Distributed networks in television automation is also here now and is also here to stay.

A Monitor Alignment Color Bar Test Signal

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A new color bar test signal is proposed for the alignment of color television monitors.

The test signal dispenses with the need to use measuring instruments for setting up hue (chroma phase), saturation (chroma gain), and black level. It calls for the existing split color bar pattern to include a thin strip of color patches and a **Pluge**[1] type signal in the lower right side black area.

Hue and saturation are set by extinguishing the red and green guns, leaving only blue light on the monitor's screen. The chroma is correctly adjusted when the four blue bars and their adjacent blue patches are equal in brightness.

Black level is set with a **Pluge** type signal consisting of small patches that are slightly blacker-than-black and slightly whiter-than-black. Black set is correct when the former is invisible, and the latter is visible.

Hue, Saturation and Black . . .

Since 1975, CBS Technology Center has had a program to study picture monitor drifts. The monitor alignment color bar test signal was developed as an outgrowth of that program.

Television monitors must be correctly adjusted to faithfully depict the quality of the pictures. Color monitors drift with time and require periodic checking, usually before each show. This is to insure that a bank of monitors in a studio are correctly matched.

This paper shall address itself to setting hue, saturation, and black level by eye.

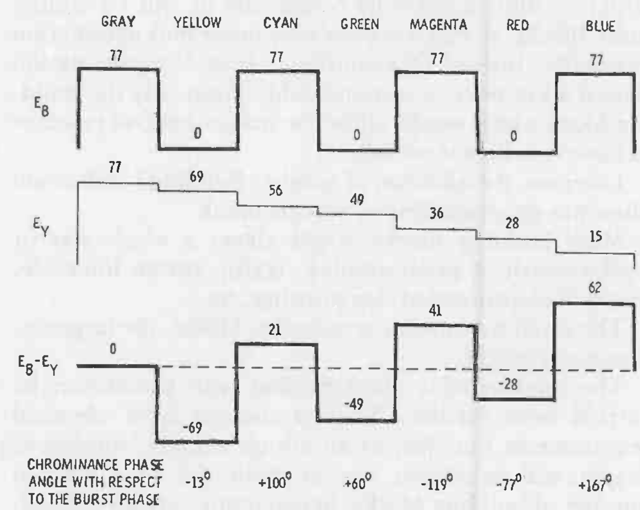

Figure 1.

Presently, a color test signal such as described in EIA Standard RS-189-A is used to set the hue and saturation of a TV monitor. With the color bars displayed on the monitor screen, the red and green guns are extinguished, thus leaving only the blue gun energized.

Figure 1 shows the waveform of the video signal, E_B, at the blue gun. The blue beam is on during the gray, cyan, magenta, and blue bars because these color bars have a blue light content. The blue beam is extinguished during the yellow, green, and red bars due to their lack of blue light content. As a result, four blue bars interspersed with black are displayed on the screen.

The blue gun signal, E_B, is the sum of the luminance, E_Y and the $E_B - E_Y$ signals, as shown in Figure 1. Saturation is correct when the two outer blue bars (actually blue components of the gray and blue color bars) are of equal brightness. Varying the chroma gain controls of the monitor changes the amplitude of the $E_B - E_Y$ signal, while the E_Y signal remains constant. Therefore, the brightness of the right outer blue bar will vary with saturation while the left outer blue bar remains constant. Saturation is correct when these two outer bars are of equal brightness.

Hue will be correct when the two inner blue bars (actually the blue component of the cyan and magenta color bars) are equal in brightness. The $E_B - E_Y$ color difference signal amplitudes are functions of the chroma demodulator phase angles in the monitor. Note in Figure 1 that the inner left bar (cyan color bar) calls for a demodulation angle of $+100°$ and the inner right bar (magenta color bar) requires $-119°$ with respect to the color burst. A change in the demodulation angle will cause opposite changes in brightness of these two inner bars. The chroma phase control is adjusted to make the inner bars equal in brightness.

Saturation adjustments will affect the inner as well as the outer bars. Therefore, the correct procedure is to first adjust the saturation and secondly to adjust the hue of the monitor.

It is difficult to visually compare the brightness of non-contiguous bars. Even when all the bars appear equally bright, chroma errors can exist if the kinescope varies in blue brightness and purity from one side of the screen to the other.

The human eye lacks sensitivity to blue light and the low luminosity compounds the matching problem. Therefore, visual adjustments of hue and saturation with the existing color bar pattern can lead to inaccurate settings.

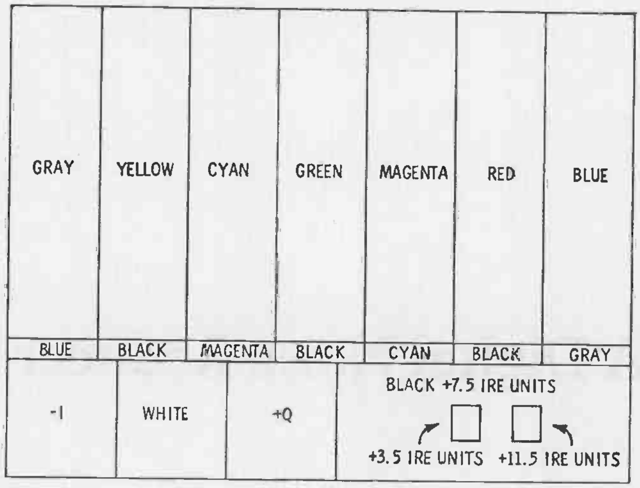

Figure 2.

A minor change in the color bar test pattern makes possible precise adjustments of hue and saturation by eye. **Figure 2** is the proposed pattern.

The solution calls for placing a small patch of blue color below the gray color bar and vice versa. Also, a patch of magenta color bar is placed below the cyan color bar and vice versa. This, in effect, places the bars to be matched adjacent to each other. Their brightnesses are precisely alike when the boundary disappears between the color bar and the patch beneath.

The patches below the yellow, green and red color bars are black. It serves to identify the waveforms of those TV lines so that they will not be mistaken for color bars.

The alignment color bar test signal also includes a **Pluge** type black set signal in the bottom right-hand portion of the raster. It consists of a slightly whiter-than-black (11.5 IRE units) patch and a slightly blacker-than-black (3.5 IRE units) patch set in a black surround. To set black level, the monitor's brightness control is adjusted so that the lighter patch is visible and the darker patch is invisible against the black surround of 7.5 IRE units.

Tests Completed . . .

The new test signals was tried for six months at the CBS Television Network facilities in New York City. Experience has proved that hue and saturation adjustments can be made by eye as accurately as with a precision photometer. Small chroma drifts are easily identified at a glance, which encourages their correction before the errors become objectionable.

The alignment color bar test signal for picture monitors has been submitted to the SMPTE TV Video Technology Committee for consideration as a SMPTE recommended practice. CBS will not file a patent application on the test signal, thereby putting it into the public domain for general use by the industry.

A Digital Noise Reducer for Encoded NTSC Signals

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Stamford, Conn.

Noise has long been a major problem for television engineers. An otherwise superb color picture can be seriously degraded by the intrusion of only a few millivolts of noise into the signal.

Until recently most random noise problems could only be reduced at the source. Thus, we have a generation of preamplifiers for cameras, tape, recorders and microwave receivers which are within a few db of their theoretical limits for signal to noise ratio. Even so, noise persists.

If an electronic journalism pickup has to be made under marginal lighting conditions, or an electronic production group finds itself five generations down from the master tape, there have been no useful techniques for improving the signal before broadcast.

Although elaborate and very expensive analogue processing systems have been demonstrated, which achieve a few db of improvement, they have not been practical for day-to-day use and have been largely reserved for special purposes such as improving the quality of pictures from the moon. This is because the improvement obtained has frequently not been sufficient to justify the expense.

The recent advent of practical digital TV technology now makes it possible to process a TV signal on an element-by-element basis with mathematical precision. In this manner certain characteristics of the TV signal can be exploited which have hitherto largely been unexplored.

The noise reduction system described in this paper uses a digital frame store operating as an adaptive recursive filter, under the control of a motion detector employing comb filtering and signal modification. Using these techniques an improvement of 15 db can readily be achieved without introducing objectional artifacts into the picture.

The Digital Noise Reducer program was initiated at CBS Laboratories in 1971, and after the restructuring of the Laboratories, has been continued as a joint development of Thomson-CSF Laboratories and the CBS Technology Center.

Principle of Operation . . .

The Digital Noise Reducer, DNR, (1) is based on a form of filtering which would be impossible to implement with analogue components. It is a one-frame, first-order recursive filter.

A recursive filter is one in which the output is a function not only of the input but also of the previous output, hence the need for the frame delay. A first-order recursive filter with a coefficient of .5 is shown in **Figure 1**.

The signals are assumed to be in digital form. The coefficient K is taken as a .5 for illustrative purposes only; in practical filters it may range from .05 to .95 or greater.

In the frequency domain it can be shown that such a filter has a comb response with a spacing of 30 HZ between the teeth and a bandwidth of the spectral lines which is a function of K. It can, therefore, be considered as a matched filter in which the spectrum of the TV signal is matched by the acceptance of the filter while random noise is largely rejected.

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*Mr. McMann delivered the paper. It was co-authored by S. Kreinik of Thomson-CSF Laboratories and J. K. Moore, A. Kaiser and J. Rossi of the CBS Technology Center, also in Stamford, Conn.

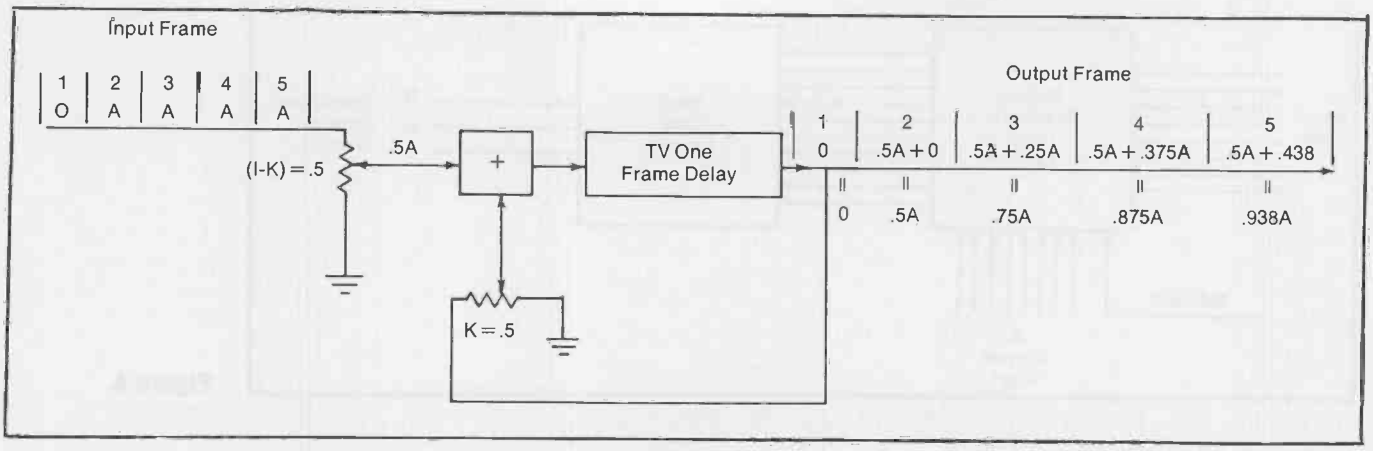


Figure 1.

The signal to noise improvement of such a filter can be shown to be $10 \log \dfrac{1 + K}{1 - K}$ and this is plotted in **Figure 2.**

A recursive filter of this type would remove noise from stationary pictures but would not be practical for broadcast use as it would produce objectionable lag on moving images. This lag, although actually a series of step functions, can be approximated by an RC type of exponential relationship.

This is shown in **Figure 3** as a function of K. It can easily be seen, referring to both Figure 2 and 3, that whenever K is chosen to give a useful noise reduction factor, lag would be greater than the human visual threshold of about one thirtieth of a second.

This difficulty is overcome by a motion detector that looks not at the entire scene but rather at each individual picture element. In other words, the motion detector

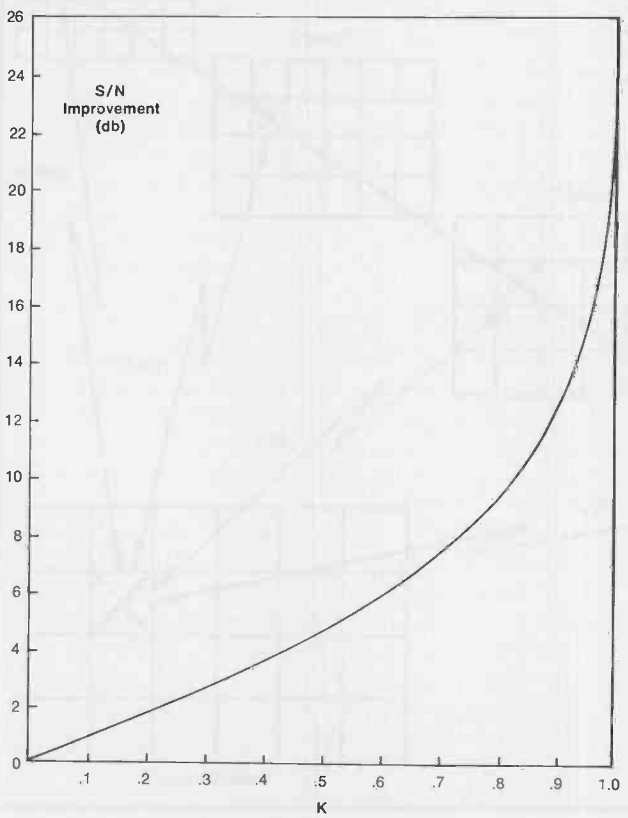


Figure 2.

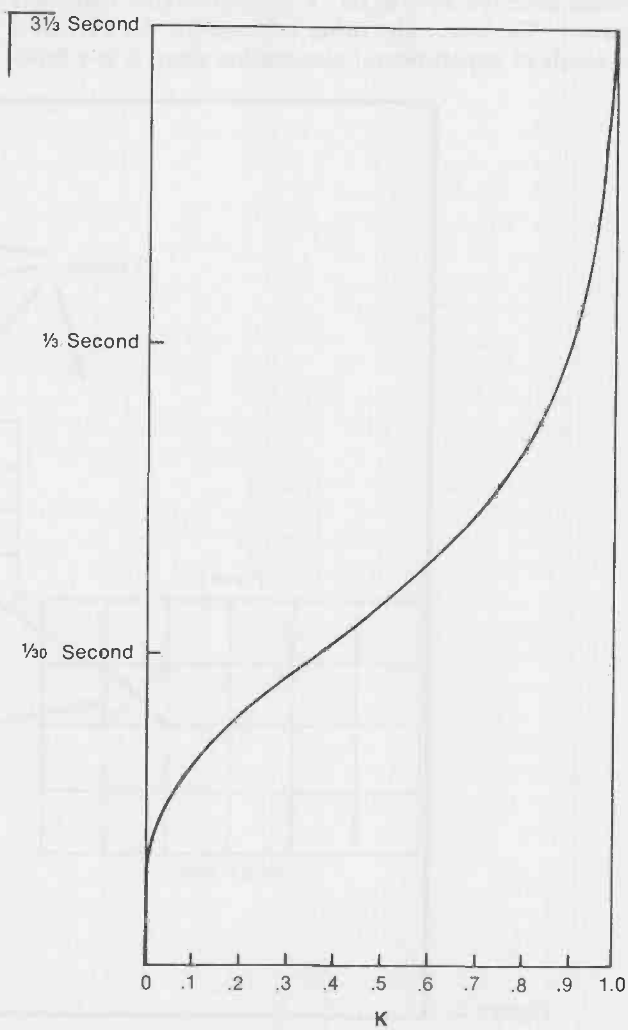


Figure 3.

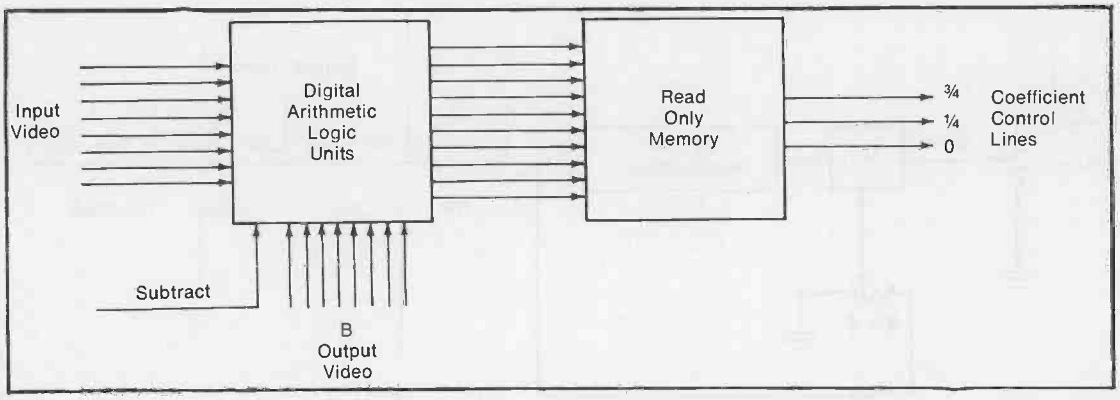


Figure 4.

makes some 358,000 decisions per TV frame, and decides whether or not motion is present at each individual element.

The **Figure 4** block diagram shows the type of motion detector logic used. The code words representing the input and output video from the recursive filter are compared in a high speed arithmetic logic unit and the coefficient K is instantaneously chosen as a function of the difference detected by PROM, a programmable read only memory. The exact algorithm selected for the PROM is the result of experimental observation since it is a function of both the noise reduction desired and of the noise originally present on the signal.

In principle, the algorithm works this way: If a large difference is detected, motion is assumed and K is selected to be zero, so the frame memory instantly refreshes itself for that picture element with new information from the input signal. In this manner lag smear is never allowed to occur. If a small difference is detected, a lack of motion is assumed and the coefficient chosen is large, thereby narrowing the bandwidth of the recursive filter for that picture element and rejecting noise.

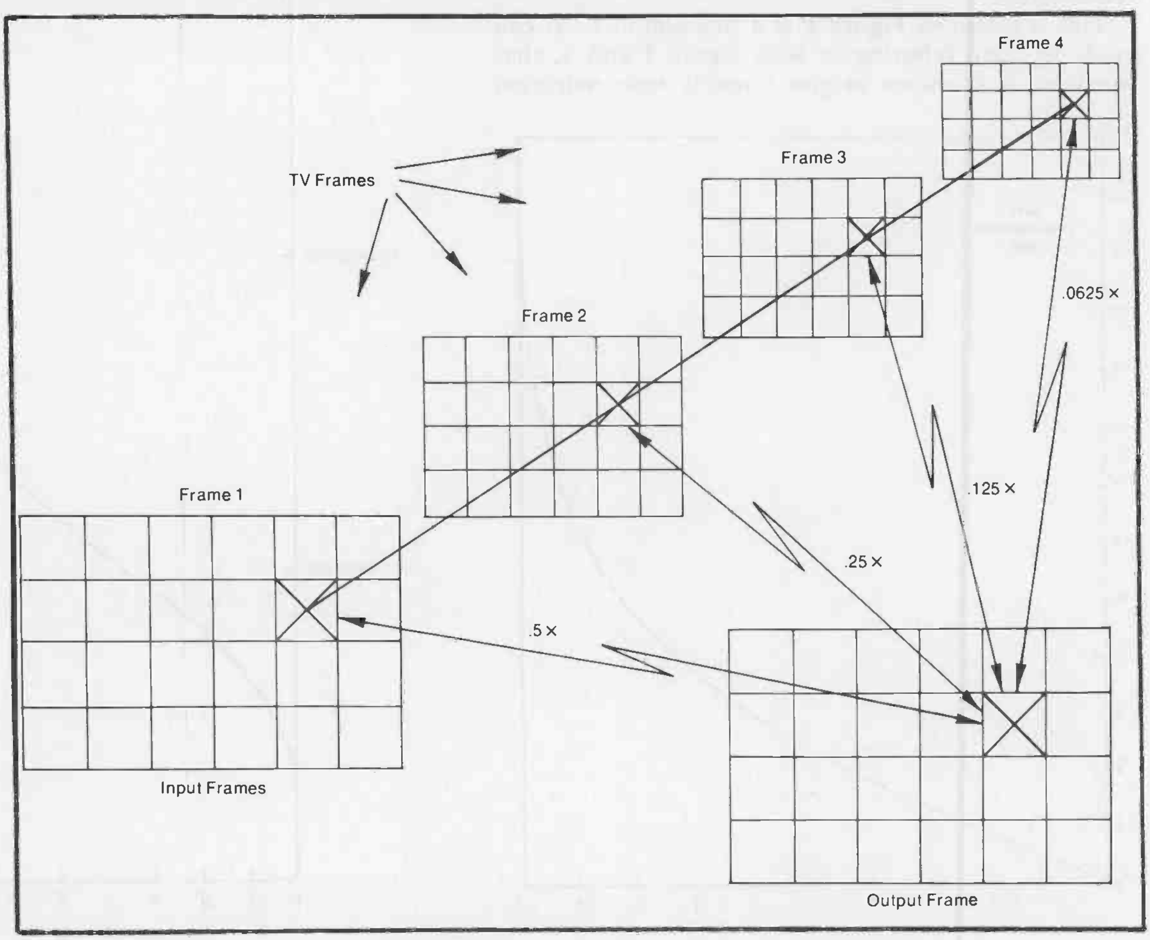


Figure 5.

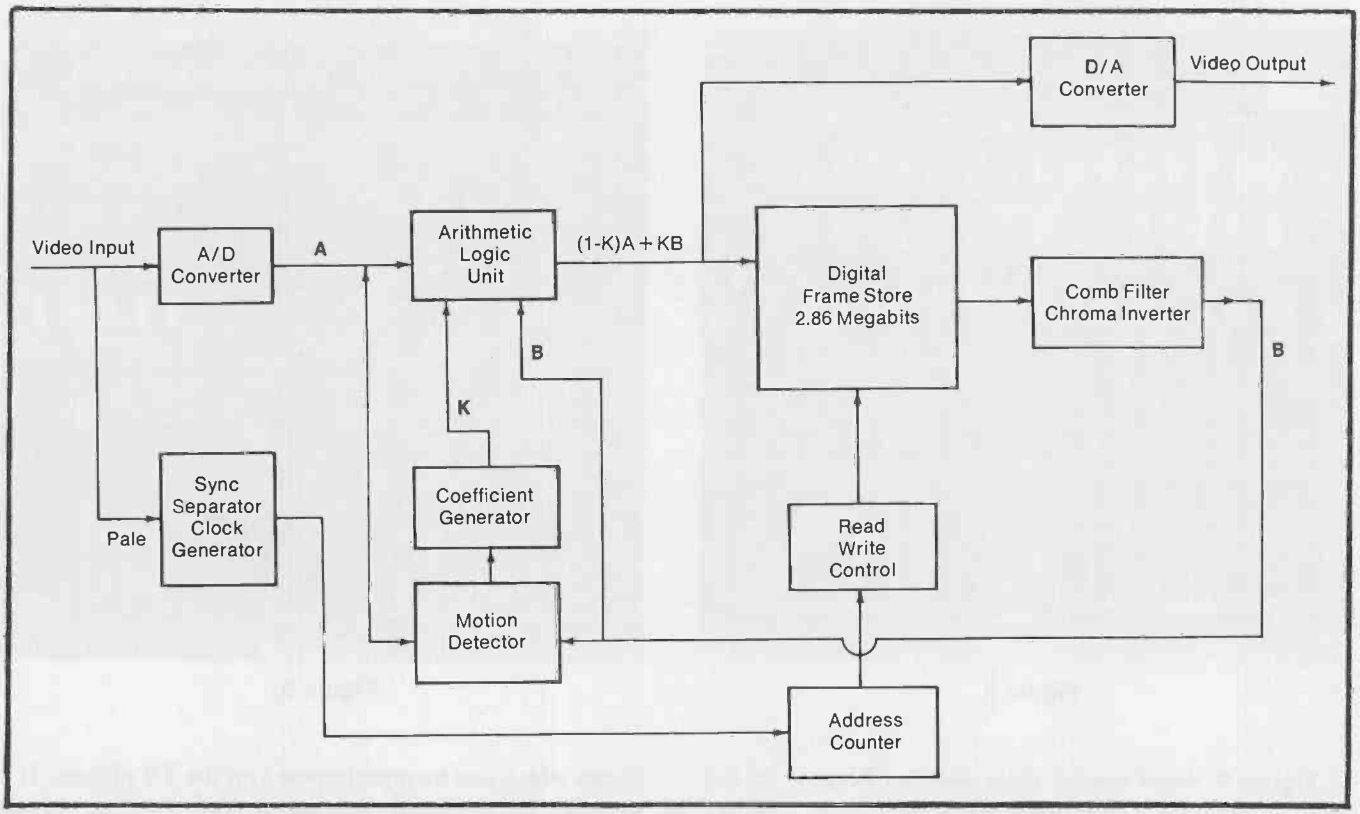


Figure 6.

One can consider that each element of the digitized picture is averaged with the same element from preceding frames with each preceding frame having a weighing coefficient which is a function of K. See **Figure 5.**

If motion is detected at any element, the averaging process is stopped to prevent lag.

The complete DNR is shown in block form in **Figure 6.**

The digital coding takes place at 3 x subcarrier using a highly accurate 8 bit A/D converter. Repeatability and accuracy are very important for this application since code words are being compared which are one frame apart and very small differences must be detected.

The output of the A/D converter is applied to a group of MC 10181 ECL high speed arithmetic logic units which generate the changing ratios of input and output signals called out by the PROM coefficient generator. Speed is important here because the arithmetic operations must keep up with the input code words which occur every 93 nanoseconds.

The resulting code words are stored in a digital frame store of approximately 2.86 megabit capacity. This store can be somewhat simpler and therefore less expensive than those used in digital synchronizers or wide window time base correctors because it is used only as a delay and does not need to have instantaneous random access.

The logic of the frame store is shown in **Figure 7.**

Input serial code words are fed into a shift register-latch-multiplexer combination and are then fed into the storage chips in parallel. The actual memory consists of 24, 16 K random access memory chips for each bit line with 48 chips on a board and 4 boards making up the complete TV frame.

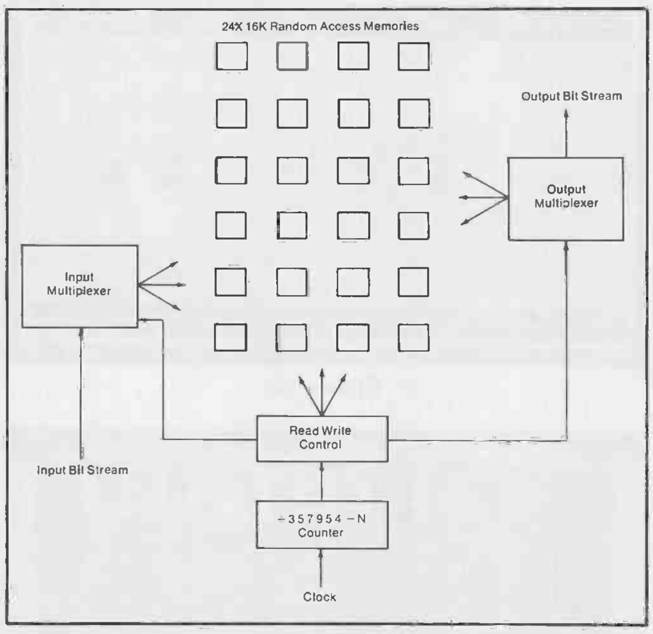


Figure 7.

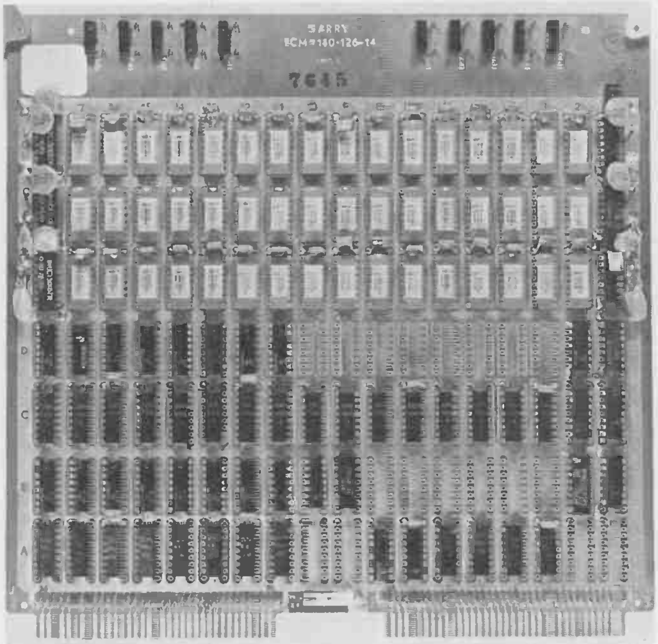


Figure 8.

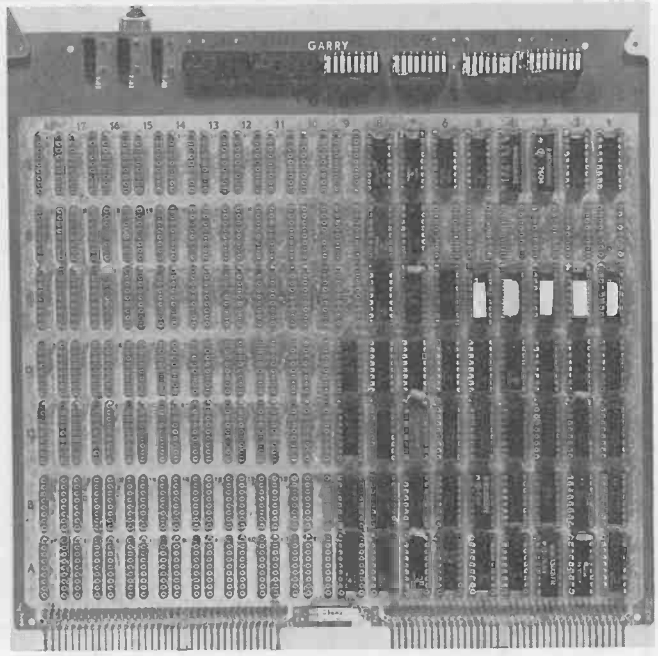


Figure 9.

Figure 8 shows one of these boards. Because of the multiplexing, each chip needs to operate only at a 500 kc clock rate so that system debugging is very easy.

The read/write control board shown in **Figure 9** has an LED display on the edge which shows the location of a cursor which can be superimposed on the TV picture. If a memory chip should develop a bad location which showed up in the picture, the cursor can be moved over the defect and the LED's will show the number of the bad board and will then locate the bad chip on the board. No oscilloscope is necessary for this operation.

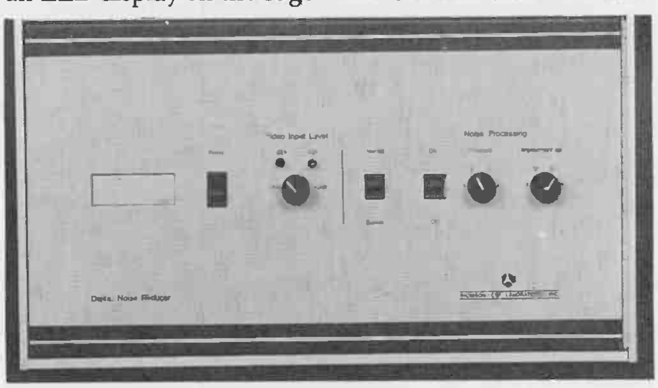


Figure 10.

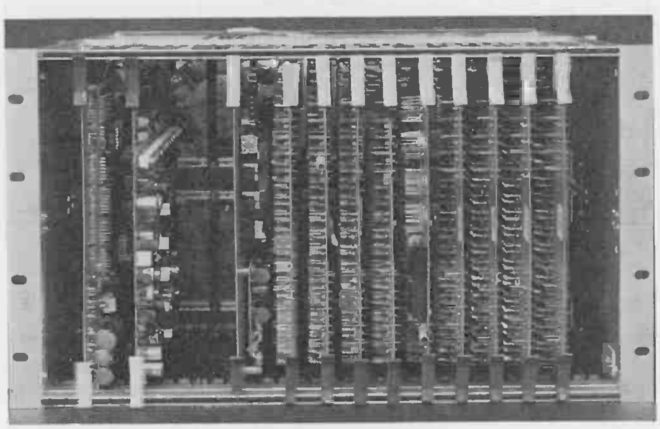


Figure 11.

The comb filter chroma inverter following the frame store is needed to invert the chroma on the picture stored in memory before applying it to the motion detector. In the NTSC system, the phase of chroma changes 180° from frame to frame and unless modified would generate a false difference in the motion detecter. Thirteen bit accuracy is needed in the chroma inverter to avoid rounding errors which would show up as spurious motion signals.

Figures 10 and 11 show the complete noise reducer housed in an 11" high cabinet. Only a single input and output video line are needed as all necessary timing signals, etc., are generated within the unit itself.

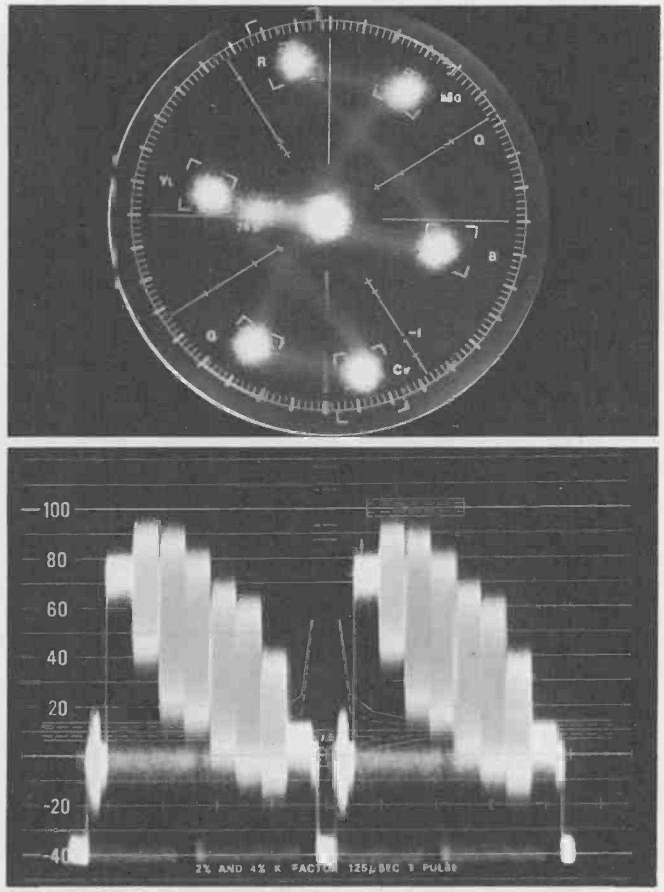
Performance . . .

The algorithm installed in the commercial version of this equipment permits 9, 12, or 15 db signal to noise improvement. The exact value chosen for a specific application depends on the characteristics of the input signal.

If the signal is very noisy and too much noise reduction is employed, the relatively quiet output picture may for an instant get disconcertingly noisy at abrubt changes of scene. On the other hand, for slightly better input signals this effect is not visible and more noise reduction can be used.

A 12 db improvement seems to be a good compromise setting for most signals. At this setting, signals down to 30-36 db S/N can be handled without any visible artifacts caused by either motion or noise.

Figure 14.



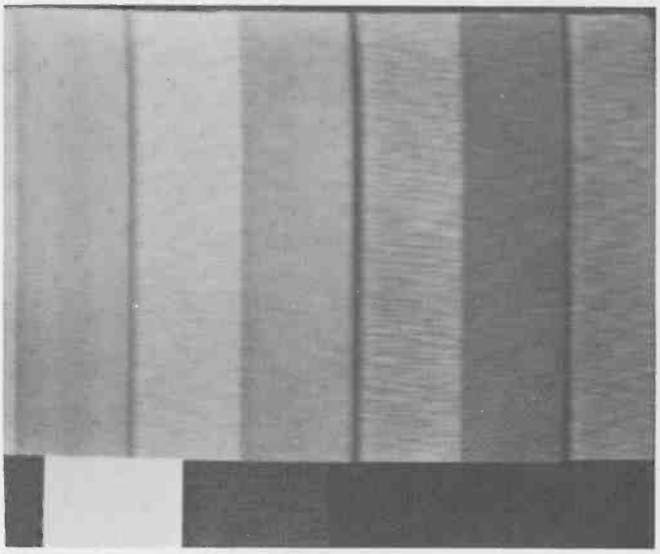


Figure 12.

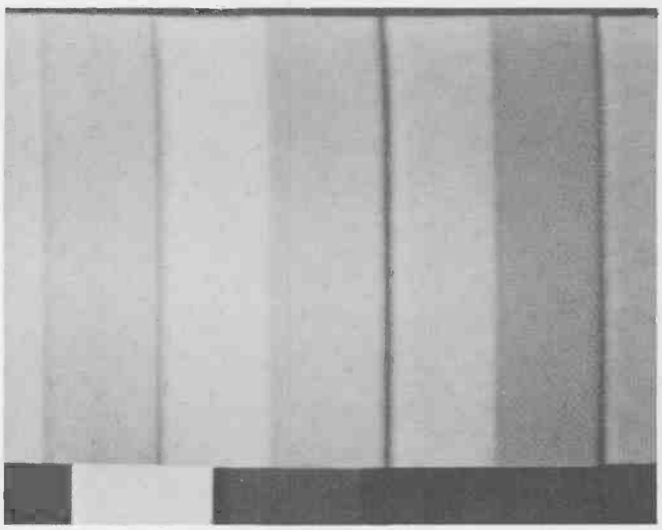


Figure 13.

Figure 12 shows a black and white display of color bars with noise typical of a multi-generation **U-Matic** ¾" tape recording.

Figure 13 shows this same signal after passing through the DNR set for 12 db of reduction. Note how the unit removes not only high frequency noise but also low frequency streaky chroma noise.

Figures 14 and **15** show Vectorscope and Oscilloscope displays of the unimposed signal and the same signal processed by the DNR noise reducer.

Still photography cannot properly show the effects of motion but skilled observers do not see any reduction in resolution or smear when the noise reduction is in operation on moving pictures.

A 40 db S/N input signal becomes 52 db at the output as measured by a Rohde and Schwartz noise meter operating in the wide band mode. A marginal 40 db

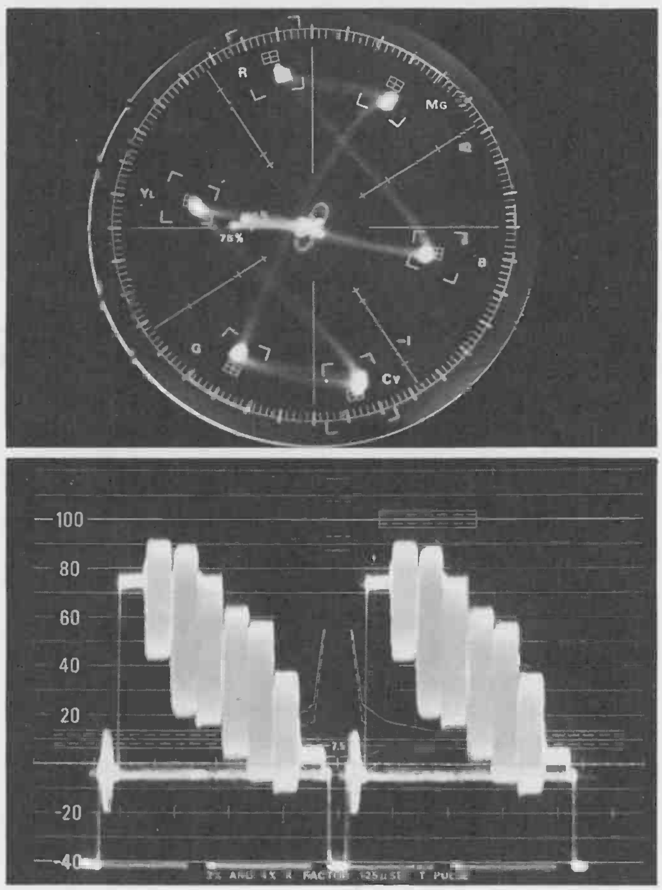


Figure 15.

Noise Reducer Applications

	Typical S/N Input	Expected S/N at Output
Electronic Journalism at low light levels	30 db	42 db
2" Multi-generation video tape	40 db	55 db
U-Matic Multi-generation	36 db	48 db
Studio Cameras-Electronic film production	50 db	56 db
Microwave transmission	40 db	55 db
CATV satellite transmission	44 db	56 db
Off air reception	38 db	50 db
Telecine film grain reduction	44 db	56 db

Figure 16.

input signal is good enough to use the 15 db improvement mode and the signal to noise then becomes a 55 db at the output; ie better than any color camera now available!

Conclusion . . .

The digital noise reduction system described in this paper gives the television engineer a new tool to improve the quality of color TV signals.

The DNR's method of operation is such that it can be left in the program line at all times; perhaps directly in front of the transmitter. It will not effect good signals but will stand ready to automatically improve poor ones.

The uses of such a device are legion. **Figure 16** lists a few of the more obvious applications.

One has only to view a noisy tape to see just how impressive a 12-15 db can be.

The fact that the DNR covers the complete spectrum from dc to 5 MHz probably accounts for the startling subjective improvement observed.

For the future, even more improvement is possible as better ways are found to employ the benefits of digital picture manipulation. Another 2-3 db should be obtainable from the present unit; bringing the total improvement up to 15-18 db and these modifications are being tried as this article is written.

In conclusion, we would like to thank Henry Mahler who conducted much of the experimental program; Dr. William Glenn, whose original invention put us on the path to success, and the management of CBS Inc. and Thomson-CFS SA, specifically, Joseph Flaherty and Michel Boxberger, who supported the development with enthusiasm.

A Portable Computer-Generated TV Titling System

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American Broadcasting Company
New York, N.Y.

The password to excitement in television production has recently come to be the word "portable". Every year, advances in technology bring the broadcaster the ability to do new kinds of production, or to reduce the expense or technical complexity of some existing feature of production.

The continuing development of solid state devices has, in the past few years, seen the transistor give way to the integrated circuit containing first a handful, then thousands of devices in a single package, allowing complex functions to be performed by miniature devices manufactured very inexpensively in large quantities.

The wide range of digital integrated circuits have made digital television picture generation possible, and one of the first devices to come into general use from this advance is the electronic character generator.

Since the switching rates of digital components have outstripped by an order of magnitude the bandwidth of the television system, and the cost of digital circuitry has dropped, electronic character generators have economically replaced cameras and slides for generating titling and logos, video signals containing basically two states: white lettering on a black background.

The character generator keyboard commands digital memory circuits storing the information of the shape of each character and the sequences of these characters to be displayed on the screen.

Such digital control capability and fast memory has given rise to further electronic video generation devices both of (1) the analog variety, such as electronic frame store devices where numerous full frames of color video are stored on a moving magnetic medium and recalled rapidly by entry of addressing information, and (2) graphic generators capable of displaying and manipulating a variety of channels of video, digitally-generated in various grey scales and colors with mixing and overlaying of channels in rapid sequence, making possible new visual effects for presentation to the television viewer.

Maps taking shape on the screen, diagrams illustrating economic trends and novel artistic effects are all within the province of these digitally controlled devices.

Computer Control of Graphics Generators . . .

The digital computer is the ultimate signal generator when used in conjunction with such digitally controlled equipment.

The computer offers memory and a facility to coordinate the needs of the user with the capabilities of the equipment, producing control sequences simply, rapidly and reproducibly by virtue of the computer's programmable nature.

Since the computer is programmable, it can be set up to adapt to the differing types of devices connected to it.

Within ABC's operations both *Vidifont* and *Chyron* electronic character generators are used. Functionally similar, these devices are somewhat different from the point of view of the operator, and material prepared on one cannot be played back on the other because the storage and formatting methods are incompatible between the different types of systems.

However, the services of a computer can be employed to minimize these problems of incompatibility.

By connecting these character generators to a computer and using the computer's memory for storing the message to be displayed, the computer can be programmed to translate the message into the appropriate impulses and control sequences required to display the message on any type of character generator, within limitations imposed only by the ability of character generators of different manufacture to display similar looking spacing, type style (font) and special characters, logos and the like.

The more sophisticated character generators on the market have the ability to create fonts and logos, and thus, when combined with a computer may be used interchangeably, giving advantages in terms of operator training, equipment availability, and incorporation of newer generation equipment as the broadcast plant develops and grows.

The computer offers other features which may be considered in application to electronic video generation.

Recent models of minicomputers economically offer storage capabilities in the range of a million characters, with tens of millions accessible rapidly from peripheral devices such as drum or disc memory systems.

Large time-shared data-processing systems may offer much more. Compared to the tape and flexible disc systems utilized by some character generators, storage capacity may be increased 10 to 1,000 times when a computer memory is used.

On a par with increased storage capacity is the ability to recall this information rapidly and *simultaneously* to two or more character generators.

In such a system it is conceivable to consider on-line fulltime storage of all titles likely to be used repeatedly, such as the names of prominent sports and political figures, or "shells" for formatted regularly changing displays such as weather reports and election results.

This computerized library, furthermore, may be updated, accessed, verified and augmented at the same time from various locations, and all locations always have access to the same data.

The computer may be programmed with the ability to keep time, and the current time recorded with the titling, so that upon recalling the message the "freshness" of the data entry can be judged. Thus if an election result, for example, is entered by one operator at one location, all locations instantly can access that same information and determine how recent the entry is.

Such a comprehensive library system merits a simple procedure for recalling information, compared to the recall "by-the-number" typical of the tape and flexible disc memories included with many character generator systems.

For example, the message "PRESIDENT CARTER" might be stored at address 100. The operator needs to know that in order to recall the title for display.

Such a numeric recall procedure is unwieldy with a large library, and the abilities of the computer can be used to relieve the situation: recall can be provided by the use of mnemonic codes.

The operator enters a code such as the first few letters of the message in order to recall the display. No restriction need be imposed by the computer on the creation of such codes, and the codes may be made up by the operator at the time the message is initially entered. Thus "PRESIDENT CARTER" could be called up by *PRES*, or *CAR* or any other combination that might be found useful.

Once such an archival titling generation system is set into operation, other services may be programmed into it.

If more than one character generator is connected, certain coordination between operations may be desirable. For example, during an election coverage, a producer may not wish to change displays "on-the-air" until he or the commentator has had a chance to preview the result to be displayed. In such a case, the computer can be programmed to recall an election display only on a character generator that is "off-the-air".

Other devices in the television plant can be controlled by the computer as well.

For example, electronic frame store devices or slide chains can be used for selecting backgrounds for the titling under computer control. Maps, pictures of individuals, symbols, or artistic material can be automatically recalled with the appropriate titling.

This type of flexibility is particularly useful in providing a distinctive appearance to election results, as well as enhancing telephone reports and interviews and weather and economic forecasts.

The computer has still more to offer with its ability to link related information and perform calculations.

Programmed sequences may be prepared for successive recall with single keystrokes without the necessity to enter the messages into the memory in the order in which they are to be recalled.

Related information may be "chained" together. The name of a reporter, for example, may be chained to a title showing the location he is reporting from and then chained to further relevant material, and all recalled simply by the operator on cue.

Computations may be performed such as computing vote percentages from election results, and automatically stored for later recall. Several displays may be linked and then updated together. Citywide vote totals, for example, can be computed from the entries for the individual precincts.

Further services also are possible, such as titling for the deaf. A series of messages can be kept on file, such as weather advisories, and then quickly and easily recalled in case a weather bulletin is broadcast.

Many of these features are incorporated in titling systems available from broadcast equipment manufacturers. At ABC, we have developed these facilities using custom equipment designed specifically for our purposes.

The Computer Interface . . .

Each model character generator has its own particular characteristics. However, digital control is common to all of them. The computer interface thus is basically simple, but generally requires a minimal amount of custom circuitry.

The circuitry will be required to match the voltage levels and polarity employed by the character generator and to generate any special timing or control signals.

Some character generators provide a special computer interface, and connection details should be available from the manufacturer. In other cases, it is necessary for the computer to substitute for the keyboard and memory unit of the character generator.

The common language used in interfacing is the eight-level teleprinter code called ASCII which utilizes eight on or off voltages or currents to represent up to 128 alphanumeric and control codes with optional error detection.

The eight voltages are typically presented to the character generator on eight pairs of wire simultaneously with a timing pulse on a separate pair, (parallel) or with the pulses presented in sequence on a single pair (serially).

Other control signals such as ready, initialize, roll or crawl control, and alarm indications may be presented as separate signals.

In some cases, more involved signalling is utilized. For example, complicated timing and control and sophisticated encoding may be employed at the interface to the character generator's storage unit (disc, tape, etc.) Computer interfacing at that point, although still involving simple voltage levels, may require more than eight data lines.

In engineering the character generator interface, there may be a tradeoff between the complexity of hardware and the difficulty of programming the computer to use the link.

No general statement can be made regarding the tradeoff, except that the merits may depend on whether the computer is a four- or eight-bit machine such as a microprocessor, or a sixteen-bit or larger machine such as a minicomputer.

Consideration is required to determine what features of the character generator will be interfaced to the computer. Roll and crawl functions, for example, may require coordination not required for regular single-page display. These added abilities thus will increase cost and complexity.

The actual programming of the link will almost certainly require a certain amount of assembly language programming. However, the overall system may be programmed either in assembler, which is a tedious process, or in a higher level language, such as *Fortran* or *Basic,* which uses more memory and may result in a slower processor operation.

The best choice for a given application will depend on the type of equipment available and the skill and experience of those doing the work.

Once the details of the link have been developed, the various applications desired for the computer system must be implemented. Many different purposes may be served, using the same equipment, merely by employing different programming or software.

For the coverage of political conventions, for example, programs allowing rapid recall of prominent names, plus the ability to add up and display the results of delegation voting is appropriate.

For election coverage, the computer software is adapted to accept automatic vote information from telephone circuits and keyboards and update and format various types of displays.

For sporting events, computation of statistics and storage of background information on players and teams is a useful feature of the software.

Taking Computer Power Into The Field . . .

Once the computer system has been developed for some of these purposes, it is useful to go into the field, to remotes, with the computer power.

Particularly suited to application of remote computation are such unrehearsed events as political conventions and large sporting events where the amount of information flowing is voluminous, and the reaction of the titling system must be rapid.

The key to computing power in the field is the data link back to the computer. Computer communication is common today, and a wide range of equipment is available from the Bell System and other common carriers as well as from independent manufacturers.

The communication normally takes place over voice-grade telephone circuits, and thus requires *modems* (*mod*ulator-*dem*odulators) to couple the DC signalling of the computer to the voice frequencies of the telephone system.

In our applications, ABC has selected the Bell System's 202S modems operating on the regular switched telephone network used by all long-distance telephones. By this choice, we maintain connection with the host computer only when required, and depend on the vast resources of the dial network to recover rapidly from circuit failures, which are rare.

Since the telephone circuit is a single pair, communication must take place serially and thus an interface may be required between the computer and the remote character generator.

The interface should also be expected to support various input/output devices facilitating communication with the computer and supporting additional system features where appropriate.

For example, a keyboard and CRT display unit may be utilized for communication. A receive-only CRT unit is useful for automatically updated material of an informational nature, such as individual election displays or such sports scoring information as leader boards in golf which may be distributed to various locations for use by talent, statisticians and so on.

The Microcomputer as an Interface Device . . .

We have selected a microprocessor-based system for a general-purpose interface. This allows us to support both *Vidifonts* and *Chyrons,* as well as other devices, by reprogramming the interface rather than by rewiring hardware.

New input/output devices may be added on to the microcomputer without difficulty, and often can be accomplished by modifying programming only.

For example, we have developed interfaces for use in our plant for connecting the computer to an *Arvin* EFS-1 as well as to a *Summagraphics* digitizing tablet. Both of these interfaces employ serial ASCII transmission, and are thus hardware compatible with the microcomputer and can be added without rewiring, simply by incorporating programming changes into the microcomputer.

The interface requires a minimum of two circuit boards, and can be packaged into a seven-inch rack opening, or small table-top unit, thus requiring little space compared to the character generator itself.

The choice of a microcomputer has allowed further sophistication of the interfacing task.

Since the system we have developed is used in conjunction with our Election Display System, we have incorporated transmission quality assurance features that will prevent display of erroneous data in the event of a transmission error on the telephone circuit. The microcomputer detects an error code bit called parity transmitted with each character of the display, as well as computing a checksum character which must agree with the checksum computed and transmitted by the host computer. The entire display is held in the microcomputer's memory until the validity of the transmission is assured in this manner, then a rapid update of the character generator's display is allowed.

In case the data fails the validity check, no automatic retransmission is initiated until a valid message is received, or until 10 transmissions have been rejected, upon which the communication attempt is abandoned and the operator is informed by the microcomputer of a circuit failure.

Using these methods, it can be conservatively projected that in the worst case a transmission error might go undetected once in a four-to-six month period of heavy daily usage.

The economic considerations of such a computer system are divided three ways: the central computer, the communication circuits, and the remote equipment.

Most costly is the central computer, with a purchase price, including software, that can range from $20,000 to $250,000 depending on capabilities chosen. Leased or time-shared computer service can reduce or eliminate the purchase expense, but processor delays beyond the user's control may impair system performance.

The communication circuits cost about 2¢ per mile per hour for long distances, more for shorter distance. Considerable variations from this figure are possible depending on distance, and total time used. A one-day installation is more expensive than one that is used for several months.

Costs can be kept down in this area under some conditions by leasing or purchasing modems from independent suppliers, rather than obtaining them from the common carrier.

The remote interface device contains only a few thousand dollars worth of hardware, and its overall cost depends on how much capability is built into it.

Our system, capable of supporting two character generators and eight keyboard terminals has a hardware price of about $10,000. The development cost for the equipment and programming was in the range of $30,000.

The selection of a microcomputer for this interfacing task resulted in a high initial cost compared to the cost of a simpler, less flexible device. We anticipate that further development and expansion, however, will be less costly because of the flexibility of the microcomputer system.

Having selected the microcomputer character generator interface opens interesting opportunities for further development exploiting the computational powers of the microcomputer.

We can anticipate using a flexible disc memory attached to the microprocessor increasing the storage capacity of the remote unit. We could then employ this memory for the purpose of storing programs for the microcomputer and material not requiring the computational power of the host computer.

This arrangement would reduce demand on the host computer as well as cut down communication line usage. Since the communication line is slow compared to an on-site disc unit, the appearance of higher speed would result from the use of a disc for those displays not requiring the host computer.

The overall evaluation of the speed of the system must be measured from the time an individual requests a display, through the time required to locate the address or name of the display and to key this information into the titling device, and for the titling device to display it.

The large memory capacity and remote communication facility of the computerized titling system results in a delay in producing the display once the address is keyed in. But this delay is compensated for because simple addressing by mnemonic codes plus quicker availability of updated information is provided by the computer's computational power.

With the storage capability of the flexible disc unit and the computational power of the microprocessor, for applications not requiring the services of a central computer, the remote system can be used in a standalone configuration.

The advantages to be gained by such an arrangement are that, with the appropriate program in the microcomputer, mnemonic addressing may be employed, with multiple entry and recall terminals accessing a common source of messages. This configuration is particularly advantageous when more than one character generator is connected to the remote interface, in that one update entry is available for recall on any of the character generators.

Conclusion . . .

Many advantages result from a portable computer facility for character generators.

They include the ability to separate the operator from the character generator itself. The operator may remain in the integration studio with the character generator remotely controlled at the field location, or the operator can be at the field location, remotely controlling the character generator at the integration point.

Mnemonic addressing of messages can add speed and simplication to the recall of titling for a fast-moving unrehearsed show.

The services of a computer for automatic updating and computation of statistics can be widely available under a variety of operational situations.

Workshops

A Workshop . . .

Broadcast Program Audio Processing Techniques

Plus, Questions and Answers

Moderator:

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Dick Schumeyer
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Jim Loupas
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Hans Schmid
American Broadcasting Co.
New York, N.Y.

John Bailie
Manager, Technical Operations
Radio Station WMAQ
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Mr. Torick:

Our topic this morning is audio signal processing. We're going to open with some brief remarks from each of our panelists, then have a short discussion period, followed by questions. Let me call on our first panelist, John Bailie.

Mr. Bailie:

When we speak of audio processing many engineers think only of magic black boxes that are cure-alls for every possible problem. Having been involved in radio broadcasting for the past 15 years, I've found few workable black boxes, and certainly no cure-alls.

Instead of looking to a single device, a systems approach to audio processing should be taken. The entire station must be taken into consideration when developing this overall plan. Every piece of equipment, both active and passive, that the audio passes through should be thought of as processing equipment.

The audio signal that is reproduced by the listener's tuner or radio usually gets its start not from the air console, but rather the turntable in the production studio. This is why it is essential to make an overall evaluation of your station's audio system.

This study should begin in the production studio, where each segment should be treated differently. Music in the form of records or tape must be handled on an individual basis.

In the case of music records, some form of special processing is usually desired. It may be equalization, compression or expansion — all depending upon the nature of the recording and the format of the particular radio station involved.

Speech may require special processing. Again, equalization. In rock radio, for example, it's exceedingly difficult to properly mix noncompressed speech over heavily-compressed music.

In the air studio we usually find ourselves with different types of audio input to our console: Microphone inputs, music inputs, usually via cart machines. You've got telephone inputs relating to call-in shows or contacts, and voice-only from network feeds. All of these may require especialized processing.

The air train, comprised of limiters, compressors, equalizers that feed the transmitter, cannot properly and efficiently process all of the varied audio sources that I have mentioned. Therefore they need to be processed individually.

As an example, the microphone input may need to be compressed, sibilance controlled, or processed in some special manner. One of the more common would be to eliminate unwanted room noise or headphone feedback.

Many examples of special processing techniques could be given, but I think we want to save as much time as possible for questions from the floor.

Mr. Loupas:

Processing in days past was almost a uniquely technical function. Today, processing is no longer necessarily nor completely within the technician's domain. It's become a function of programming and engineering.

Probably the most confusing and most difficult problem I find in the field is that while everyone, almost without exception, wants to process, there are people who process and don't know why they process.

So to that end, I would suggest that when you're talking in the new language of processing, which turns out to be Total Survey Areas (TSA); Quarter Hour Maintenance (QHM); and Aural Dominance of a Market, you're in a whole new idiom. It requires the cooperation of your program director, and your chief engineer — which sometimes is very difficult.

The point I'm making is to know **why** you're processing.

Hopefully, this panel will not only be able to help you know how to process, but to **know why your processing.**

In essence, that means you must analyze your market, because your market and your processing will be unique; it will be one of a kind. There will never be another one like it.

You must analyze your market and determine who your primary competitor is. He's the first guy you have to go after.

Finally, design and specify a processing system to serve your needs. It must be a system that you understand, one that you can control, one that does not control you.

Mr. Schmid:

I'm here as a broadcaster, and as a broadcaster, to slightly exaggerate, I don't believe in processing amplifiers whatsoever in the broadcasting plant. I cannot help thinking that what we have here is the symptom of providing rubber gloves for leaky fountain pens.

You can process audio in several ways — volume-limiting, compression, expansion, loudness control, frequency or presence enhancement, etc.

Let me quote from a recent forum on compressors and limiters:

"Ask any audio engineer who uses limiters and compressors just what he wants and has in the way of compression and limiting equipment, and you will get as many answers as you have engineers. Yet modern broadcast requirements require the use of these components and the need for more versatile componentry is required. Thus the spiral of better equipment and greater confusion moves on."

I think I have done enough ad-libbing. Maybe I'll have something else to say later on when we come to program level.

Mr. Schumeyer:

I suppose in any discussion of processing you have to go back to the basics and figure out what you want to do and why you want to do it. But the beginning has to be back in the production room, where it all starts.

The air chain is not going to cure sloppy production techniques. It may hide some of them, but it's not going to make you sound much better.

If you've already created distortion in the turntable, you're not going to clean it up with some magic black box in the air chain. So the idea is to make sure the system is flat. And consider it as a system, not just a bunch of black boxes.

I think one of the problems I've found most predominant is a lack of headroom and consideration of system levels. In most stations you'll walk in and there'll either be no headroom or the levels will be all over the place. And that's one sure way of getting yourself into trouble.

You can be clean all the way through and it might look pretty good with tone. But once you start putting program material in at 10 or 15-16 db above the tones, you wonder why it doesn't sound like the tones did.

My thrust is to get back to the basics; look at everything, consider it as a part of the system. Step back and look at all these black boxes and figure out exactly what it is they're doing and go from there.

Mr. Small:

I want to talk about what I see as the future of audio processing. Given the rapid growth in digital processing, I think it's going to be upon us very quickly.

Those of you who are involved in video are aware of the revolution that's been taking place in the last two years in the digital processing of video signals.

Currently, there is some digital going into audio. Usually it's simple delay lines, either pure digital, where there's an A to D conversion done, then a shift register, then a D to A conversion, and charge a couple of devices, which are kind of hybrid digital on analog systems.

I think we're moving in the direction of pure digital signal processing, where all the work is done in a digital environment. We may even have digital studios some day.

A number of people including the Japanese are working on digital audio tape recorders. I think there's one being commercially marketed now in this country. Some of them work around modified videotape recorders.

The point is that once it's digitized, the opportunity for doing extremely sophisticated processing is definitely available.

There's a professor in Utah who has taken old Caruso recordings in very poor shape and, after cleaning them up, has been able to enhance them. He was able to put information onto the recordings that wasn't there, by making certain assumptions about the human voice. He wound up with tapes of Caruso that were probably better than the original recordings.

I think the problems of broadcast signal processing, even our most sophisticated problems of variation from transmitter to transmitter, are probably trivial in comparison with this kind of decision-making. I think digital processing would allow you to handle individual situations. Transmitter tilt, transmitter balance, all the real-world problems could be dealt with by this kind of system. And I think it's in the cards and it may not be that far away.

Mr. Williams:

So far we've talked about the entire audio chain from production room to transmitter. I'd like to talk about why the material that comes out of the recording studios, doesn't sound the same way on the air as it does in their studio.

I've tried to justify this, in part in the past, by saying by the time you get the material into the average listener's home, some things have subtracted from the quality of the original program.

We ran a set of response curves on various types of commonly available portable and high fidelity receivers to show some of the limitations of these devices. For example, we measured the response of a little Wollensak 45-15-FM mono cassette recorder/portable radio. As you can imagine the response is quite limited.

Why does FM transmissions in automobiles and FM transmission in portables sound so muddy and so terrible? It's because the receivers are not very stable.

Some use a very fast AFC circuit to overcome this at the low end. The AFC is actually chasing the audio signal and cancelling it out. In the higher end, they put in some kind of a very quick filter to get rid of the stereo pilot and not to cause any more distortion in the receiver.

A Sony CS-550 AM-FM stereo cassette portable has almost a reverse response curve, as far as, you might say, the lows to the highs. The same problem in the bottom end, but a much better filter in the top end.

In a Marantz, hi-fi home-type AM receiver, the bottom end held fairly even though the AFC circuit started to chase the low-frequency information. There was not much in the way of high-end response in this receiver.

What I'm trying to point out is that a lot of the processing that people are attempting to come up with today is to overcome the response limitations of receivers.

Automobile receivers that we've run curves on, made by either foreign manufacturers or domestic manufacturers, all follow the same type of a bell-shaped curve. They tilt one way or another as one manufacturer or another has tried to brighten this up or boost this up or put more kick behind the padded dash.

I think we really need to say that in the future, audio processing should go much further, especially in AM radio. We're going to have to see some new research and development in the technology of AM reception.

FM has gotten all the attention all these years. Hopefully, with the advent of AM stereo, we're going to see some decent technology applied to AM stereo receivers that will give us that kind of bandwidth on AM.

Mr. Torick:

Well, we've heard from our panelists. We'd like to hear from our audience now. Are there any questions?

Question:

I'd like to first ask Mr. Williams where those receiver responses were measured? At the speaker? At the IF? Where did you make those measurements?

Mr. Williams:

That particular set was taken at the output of the amplifier, and we had to disconnect the loudspeakers, particularly because the resonance of the speakers in their cabinets would cause all kinds of other problems.

We did make some third-octave measurements in some car radios with a bar-graph display. These measurements show you things like the resonances of automobile dashes, the speakers and the door panels that produce some very wild-looking things as far as the base region.

Question:

Why has the trend in broadcasting seemed to be sacrificing of fidelity of reproduction for compressed, processed loudness?

Mr. Loupas:

I think we, as an industry, have created a mentality that our material must be louder than anybody else — "want to be louder than the guy across the street."

That's the first thing one hears.

The next thing is "I want to be bigger than the guy across the street." It's a very serious problem, trying to compromise between the requirements, in most instances, of a program director, and some elements of good taste.

And they are compromises. Everything we're talking about, in general, in terms of actual processing, has to do with good taste, because it's very subjective. Particularly in AM, you're talking about some very subjective adjustments.

In terms of FM, notwithstanding the small receiver, you're talking about a full bandwidth system. In AM you're talking about a limited bandwidth system. So as a result, with no standards, you're compromising.

Questioner:

I'd like to define fidelity. Fidelity, to me, is the transmission of the program as it originally started at the beginning of the system, with no change in any of the attributes that we might think of.

Mr. Schmid:

The gentlemen who defined fidelity was clear to say that it's fidelity "as he defines it." The answer to the fidelity he wants is, of course, a theater ticket to the concert hall.

As engineers we must understand that for a given goal that you have picked a medium to get that across. You did not buy the ticket to the concert hall. You got it through a pair of wires or through the air, and that imposes certain limitations. That is where we lack standards, which is, of course, a great hobbyhorse with me.

Mr. Williams:

I'm in violent disagreement with the receiver manufacturers, as far as roll-off is concerned. Since most every AM receiver is like that — and there are some notable exceptions — I don't know what the solution is. I also anticipate some difficulties in relation to AM stereo because of the emphasis curve. I've heard arguments both ways on that.

When a client asks me, "What am I going to do about AM stereo? I say, 'I don't know.' " Because, as we're all aware, those radios start falling off at 1000-2000-3000 cycles. We're dealing with what almost appears to be FM de-emphasis in a non-standard manner. I don't know what the solution is. I think the solution is much more complex than something I can just come off the top of my head and answer.

Questioner:

The reason why I asked the question is that I feel pre-emphasis in any system, AM or FM, is not necessarily desirable. This is the point in the system, the transmitter, where you have discrete limits on what you can do. You can't exceed the 100% modulation on AM, you can't exceed the surface deviation on FM. And the minute you put pre-emphasis in the system, you've bought yourself a whole bag of worms.

Mr. Torick:

Well, as all the AM people in the room are aware, crank the high end up in your AM system and see what happens to your bandwidth, especially if you've got a funny antenna.

Mr. Small:

I think a very critical point was just touched on when the limitation of 100% modulation was mentioned.

Audio, especially high-quality audio, as we know it, is a very, random program material. It's a very random signal by engineering standards. In fact, most analysis of audio is carried on using the techniques that are used for analysis of random noise. And one of the requirements of a good audio system is the ability to deal with a very wide range of amplitude and frequency-type signal.

Along comes an AM transmitter that has a limitation of 100% negative modulation, which is quite absolute. You can't do anything more than turn it off.

But in FM, your 100% modulation definition is extremely arbitrary. It's just that the FCC decided: "There it is." There's a fair amount of confusion over the difference between occupied bandwidth and peak deviation.

A recent paper by Richard Tell of the Environmental Protection Agency that appeared in IEEE Broadcast Transactions reports on a statistical analysis of occupied bandwidth of FM stations in the Washington, D.C. area.

The report contains some very startling figures, which raised the question of the relationship between the FCC's very absolute definition, which has become even tighter now, and the new ATS modulation rules and occupied bandwidth. When you start talking about occupied bandwidth, you start talking about the basis for the bandwidth allocation of protection ratio. One of the points the article raises is that there is no clear relationship between limits on modulation, occupied bandwidth, and the protection ratios.

The point of all this is that the monster of heavy processing has been created by the existence of this absolute limit of modulation. It's really the difference between a classical music station which, if it abides by the rules and doesn't want to process, is going to sound 8 to 12 dbs below a moderately-processed contemporary music station.

If they are allowed to sound the same volume, with much less processing, and the occasional overdeviations of the classical music station were accepted, there really wouldn't be any problem.

Mr. Torick:

Eric, I think you're referring to the new rules, which, as I understand, haven't come out, but we'll describe modulation, negative modulation limits for AM in terms of over-modulation peaks per unit time.

Mr. Small:

There's a five-millisecond window. Any occurrence of modulation within a five-millisecond interval is considered to be one occurrence, and you are allowed 10 of those occurrences per minute. If it exceeds 10, that is overmodulation. The ATS rules are a little more involved than that, and it's my understanding that those rules will soon be applied to FM modulation in general and negative AM modulation.

Mr. Torick:

These limits are primarily, as I understand, for the purpose of automatic transmission equipment. Nobody really expects to sit there counting flashes of that lamp.

Question:

A problem that's bothered me for quite a few years is the tilt and balance that we see in transmitters. This is expensive to get rid of, especially if you don't have one of the new highly-efficient transmitters. Is there anybody that's doing any processing to compensate for this problem particularly in AM?

Mr. Schumeyer:

With the older transmitters and the power-supply balance problems which have been discussed in numerous papers in the last few years since the advent of the PDM and the other more-transparent, high-efficient transmitters, there isn't a whole lot you can do.

You can try beefing up your power supply, but then that'll create other problems.

You've still got that hugh chunk of iron in there for the modulation transformer, and that's one of the biggest limiting factors you've got, and it's awful tough to get rid of that.

Some of the older transmitters, were never designed to modulate more than about 30 to 40 percent, on the average. And when you start highly processing or heavily processing and getting the average up to 50 or 60 percent, it just doesn't want to do it.

I don't really know the answers. I've been wrestling with that one, too.

You can process and you can sound clean and nice and pretty and loud, but there's always going to be some kind of a trade-off if you're trying to make a Model T run 70 miles an hour. It was never designed for it.

Question:

Is there anything that can be done in a system to increase your output level without compromising the fidelity of the signal? Are there any things that are being done wrong now?

Mr. Torick:

Peak program meters help somewhat. If you've been measuring what goes into the transmitter by looking at sine waves and setting up your compressors and limiters that way, you've got to allow about a 10-db headroom and back them off some. Peak meters help quite a bit, because they'll give you a good idea of what really is happening with program material.

Questioner:

What I'm saying is, is processing always necessary to get that extra sound?

Mr. Torick:

You don't have to process, no. It depends on what you want to do. If you're going to process, there are some very good ways of doing it and there are some very bad ways of doing it. Overprocessing is worse than no processing at all.

Mr. Bailie:

The audio processing that we're using today really is not all that destructive. What is more destructive is the little things that get passed over at radio stations in your production studio.

Headroom is so very, very critical. There are more problems involving clipping, which is, you know, your worst form of possible distortion.

I don't think when we hear a radio station or an audio signal that really sounds bad, it's necessarily due to the processing techniques that we're using today, if they were applied correctly. Audio can be overprocessed. It also can be underprocessed.

If you use no audio processing, if you just play the record through the console directly to the transmitter, your coverage area would be greatly reduced. You would really be doing a disservice to the majority of your listening audience, no matter what format you're running. Even with a beautiful music formats on FM, you require something to improve the basic signal.

Panel Member:

Audio processing doesn't have to include hard clipping. You can process quite a bit and still sound clean. I think that's what we're all talking about: the proper application of the newer processing gadgets that are available today.

Here's one problem I've run into several times. You find a station that's put in a new transmitter, and doesn't like the sound of it because suddenly it's passing all of the junk that the older transmitter chopped off, junk that you created back in the production room.

That's why I'm a big fan of going right back to the beginning and analyzing every piece of equipment and then putting it all together and knowing exactly what each piece contributes to the overall effect that you're trying to get to.

Questioner:

I think, in general, most of us skirt this issue because we're afraid of showing our ignorance. And as a result of that, a lot of people are using a lot of unnecessary processing. As a result, on-the-air sounds are becoming more and more horrendous.

I think it would be of great value if somebody would address themselves to the amount of limiting, compression, expansion, clipping, what-all, that should be used in a system and the proper ways of setting it up and calibrating it.

Mr. Williams:

First, I'm very glad you brought up that particular comment, because we get a great deal of calls from the field. People will call and say, "Yeah, it sounds great, but the meter just isn't hanging like I anticipated it would."

Gentlemen, mankind can never listen with his eyes.

You've got to listen with your ears. Meter readings don't mean a thing when it sounds lousy. And if it sounds great, to quote what the people in the recording industry tell us: "No matter what it does, if it sounds great, that's what they're looking for."

As far as how to use the equipment, that is a big problem.

We sent out limiters, Eric sends out limiters, CBS or Thomson-CSF sends out limiters on evaluation. One guy calls back and says, "It's the greatest thing since bottled beer." The next guy says, "Boy, I wouldn't give that thing to my worst enemy." It's the same damn box.

There is a lot of misunderstanding on what these devices are to do. There's a lot of misunderstanding on how to test them.

We write our manuals hoping that people can put them on a bench and if they follow step one, two, three, four, and have the proper tools and test equipment, they can go through the device and figure out what it is doing and everything else.

Perhaps we more than some others have put more controls on our equipment so the engineer and his general manager or program director can tweak and tune and try to get the degree or apparent compression or lack of it than other manufacturers.

Life was simple when you had an input control and you adjusted a meter until it was somewhere between the green and red range, and the output control till the blinker went on just enough that you said, "Is that occasional enough to keep the inspector off my back?" We now have devices where you've got trim pots all over the place.

It's amazing how many people will get this type of equipment and put it on the bench, and just look at it and study it, then put it in the rack and hook the thing up and say, "Yeah, that's great" or "No, that's lousy." In the latter case, back in the box and off it goes.

The device should go into the production room, installed between the turntable pre-amp, and the mod amplifier. Sit and play a selection of records, tweak, and get a feeling for what the thing is doing. Learn to listen to a particular piece of equipment.

We go into some stations that have let their audio processing equipment build very much like an octupus.

You walk in the production studio, and here is a Level Devil on the production mike, followed by a Leveling Amplifier, followed by a Gain Brain, followed by a Kepex. Then out of the board it's going into somebody's stereo compressor on its way to a stereo cart machine, because the cart machine doesn't have any headroom and the jock doesn't watch his level. After all, the meter's been broken for six weeks.

Mr. Loupas:

Amplifying on Jack's remarks, it gets back to my original remark about knowing the equipment you're using.

Unfortunately, all the committees in the world could never sit down and give you a specification for the amount of compression or the amount of clipping or the amount of expansion, or whatever.

Each piece of equipment is unique unto itself and the manufacturer is really the only man you can look to. It comes down to getting that piece of equipment and knowing it. Know what it's going to do.

Typically, you put a piece of equipment in the transmitter facility, you've got blower motors going, you're sitting out there trying to adjust it. You can't run out to your car because the car radio's got a window antenna. Or it's being overdriven, and where are you going to find a reference?

Number one, you have to find a reference; and then you have to understand what you're doing. If you turn that screw a quarter of a turn, you'd better have some idea of what it's doing on the air because you're going to be accountable for what that's doing.

It all comes back to the same basic philosophy that everyone at this table has mentioned: If you put garbage in, you get garbage out. And the proof of performance is the minimum required.

Mr. Williams:

Concert hall realism is a nice thing that hi-fi salesmen like to talk about. But the modern record played on a radio station in the United States today, unless it is purely classical and more than likely done in Europe, is not a concert hall rendition.

Modern recording techniques with 24-track recorders, or even two 24-track recorders synchronized together, will have musicians on a single cut, in a single record that maybe have never even met each other, that came from diverse countries, done in diverse studios.

What you listen to when you listen to these records are an illusion that the producer, the arranger, and the engineer mixed onto that record. It is an illusion. It is not an original recording.

What we're trying to do is process it into the illusion that you, as a broadcaster, are trying to represent to your listening public. And because every artist is different and every arranger is different and every mixer is different, we do get a wide variation in quality of discs.

Take for example, the disc that won the Grammy for Stevie Wonder, the "Songs in the Key of Life" album. It's musically excellent and it's sonically dull. It's because where they mixed the thing, they had super-bright speakers, since there's not an awful lot of top end on it. When you try to mix that record into your program format the horns are just plain dull, compared to your other programming.

That is the kind of source material we're talking about back in the production that needs to be processed and brought up to life to provide a sonic balance with the other things you're playing. Then you have created the illusion at the listener's end that is more compatible to the other material you're playing.

Questioner:

The fidelity I was talking about was not necessarily true-to-life. I'm talking about what we, as broadcasters, are given as program material, whether it be on a disc, on a tape from an ad agency. Whatever we are given as material, we should try to reproduce exactly what came to us.

Mr. Williams:

In other words, if it comes to the door dull, it should go out on the transmitter dull.

Questioner:

Sure.

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Mr. Torick:

Gentlemen, I regret that the hour's gone. I want to thank our panel this morning for a stimulating discussion, and invite you all to continue enjoying the activities of the convention.

Thank you.

AM Stereo Workshop

Plus, Questions and Answers

Moderator:

Christopher Payne
Assistant to the Vice President for Engineering
National Association of Broadcasters

Participants:

Mike Davis
Thomson-CSF Laboratories, Inc.
Stamford, Conn.

Leonard Kahn
Kahn Communications, Inc.
Freeport, N.Y.

Harold Kassens
A. D. Ring and Associates
Washington, D.C.

Al Kelsch
Magnavox Corporation
Ft. Wayne, Ind.

Arno Meyer
Belar Electronics
Devon, Pa.

Norm Parker
Motorola, Inc.
Schaumburg, Ill.

Mr. Payne:

Our subject this morning is AM Stereo.

We have with us the four major proponents of AM stereo in the country. We've given them a double assignment, not only to talk about their system and their particular viewpoints on the status of technology of AM stereo, but also to speak on a particular special area of AM stereo so that everybody can be brought up-to-date.

Let me first introduce the panel: **Mike Davis**, Thomson-CSF Laboratories, Inc., Stamford, Conn.; **Leonard Kahn**, Kahn Communications, Inc., Freeport, N.Y.; **Harold Kassens**, Chairman, National AM Stereophonic Committee and a partner in the firm of A. D. Ring and Associates, Washington, D.C.; **Al Kelsch**, Magnavox Corporation, Ft. Wayne, Ind.; **Arno Meyer**, Belar Electronics, Devon, Pa., and **Norm Parker**, Motorola, Inc., Schaumburg, Ill.

Mr. Parker will talk about the similarities and the differences between the systems.

Mr. Kahn will talk about transmitter interfacing between AM stereo equipment and your transmitter.

Mr. Meyer will discuss monitoring and AM Stereo monitors.

Mr. Davis will talk about the special problems with limiters and processing devices for AM stereo.

Mr. Kelsch will discuss the future AM stereo receiver.

And Mr. Kassens is going to talk about the activities of the AM Stereo Committee and some of the proceedings before the FCC.

You should understand that these people have competing systems and AM stereo is very competitive. So after their special assignments, they're invited to speak on the advantages and disadvantages of the different systems. We will begin with Norm Parker.

Mr. Parker:

I'm going to be talking about all systems. So I'd like to start off by saying that any remarks I make are not intended to be a quality comparison of the systems but merely to point out various similarities that do exist between all the systems.

There are four systems and they're all good systems and they all work. From that point on, I'll begin to talk about the four systems and their comparisons.

It was easy to add stereo on FM because there was a lot of spectrum space that was unused. It's not so easy to put stereo on AM.

We are fortunate, however, that the earliest methods of modulation were double-sideband signals. Since they were double-sideband, it's possible to put additional information in, since we have both an upper and lower sideband present.

In a normal double-sideband transmitter, the most important characteristic is that the sidebands are symmetrical about the carrier and each are identical in phase and amplitude. Needless to say, there's a considerable amount of redundant information there.

In all the proposed AM stereo systems each proponent makes use of this redundancy to change the phase and/or amplitude of the relationship between the upper and lower sidebands to generate the stereo signal. The way in which we change the symmetry of the sidebands will indicate the difference between the signals.

As in FM stereo the transmitter must be modulated in such a way that there is a compatible signal available to the monophonic receiver. That, of course, is the left-plus-right signal.

The Belar system, which was originally submitted by RCA, is a modified signal, L + R, that amplitude modulates the cosine term and results in an AM envelope. In other words, the envelope of that resulting signal to an envelope detector is clearly L + R.

A common characteristic of all four systems is that they modulate only the phase angle of the carrier. If you don't do that, you couldn't have a compatible envelope.

So the comparison between all these systems will be in the way they modulate the phase angle of the carrier.

In the Belar system, the choice is FM for the carrier. In other words, the carrier signal inside an envelope will have the zero crossings wobble back and forth in such a way that additional stereo information is present without changing the shape of the envelope. That's the key characteristic.

This is a combination FM system with pre-emphasis. Pre-emphasis starts at F-Zero, which is about 1600 cycles which means that the output of the transmitter is phase modulated above 1600 cycles and frequency-modulated below 1600 cycles. The maximum phase deviation on the transmitter is plus-or-minus 1250 cycles. Which means that it's several radians of phase modulated at two or three hundred cycles.

A Common characteristic of this signal is that the sidebands are asymmetrical for a simple signal. In other words, if we put in an L or an R, the sidebands are symmetrical.

If F over F-zero was not there and we had a modulation at 1250 cycles, the system would become a single-sideband system at one frequency; and at other frequencies, of course, the symmetry of the sideband changes. So the symmetry of the sidebands is a function of frequency.

In Mr. Kahn's system the phase angle is modulated in a slightly different way. In this one, the phase of the audio information is shifted 90 degrees so the upper and lower sidebands cancel one another.

This is a unique situation that occurs when the phase angle of the carrier is adjusted to match the AM sidebands. In other words, the Bessel functions of the first order, which are the first sideband functions, are linear up to nearly pi/2. So the first-order sidebands contract and cancel on either side of the carrier.

In this particular case, for simple modulating signals, the left signal is on the lower sideband and the right signal is on the upper sideband. This then becomes a compatible single sideband signal, compatible in that the envelope is always the envelope that we desire to have.

The next system, Magnavox, uses essentially the same kind of phase modulation of the carrier. The difference information is not shifted by pi/2, and so the result is that the sidebands are symmetrical.

The upper and lower sidebands have the same magnitudes; they're symmetrical about the carrier in magnitude, but not necessarily in phase. They're shifted slightly in phase, which means that the resulting modulation is at a slight angle to the carrier.

This system has one other thing that none of the other systems have—a pilot carrier included at approximately five hertz which can be used as a stereo indicator.

Basically, this system is one in which the phase angle is modulated to produce the stereo difference system. The envelope, again, is compatible.

The last system was developed by Motorola and is basically a quadrature system, wherein the transmitted signal is the arc-tangent of the radio of L − R to 1 + L + R.

At the receiver it's turned back into pure quadrature.

Here again, you can see that the similarity in all the systems is that the phase angle carries the stereo information, and the envelope carries the monophonic information.

Mr. Payne:

Our next panelist is Leonard Kahn, who will discuss interfacing between stereo encoders or stereo adapters, and transmitting equipment.

Mr. Kahn:

As Mr. Parker indicated, there are two degrees of freedom in modulating a sine wave. One is to vary its strength and the other is to angularly modulate. In broadcasting you've been working with one degree of freedom, amplitude modulation and your transmitters therefore, are well-designed for this technique.

The second requirement is to get angular modulation through the transmitter—which it wasn't designed for. Fortunately, all transmitters do an efficient job of passing RF through the various stages of amplification. However, the antenna coupling could present a problem.

As you well know, if you had extremely high Q's most of the power would stay in the tank circuit or the LC circuit and just circulate. It wouldn't go any place. So there is a ratio of loaded Q to unloaded Q which is a requirement for good efficiency in a transmitter design.

Why is this important? Obviously, manufacturers build the coils and capacitors in your transmitter with the best material they can. That's why those tank coils are so big. They then must load those circuits heavily so that this relationship works out to a decent factor.

For example, if you had an unloaded Q of 200 here and a loaded Q of 5 you'd end up with 97% efficiency. That is a requirement. You definitely want to make those loaded Q's as low as you can and the unloaded Q's as high.

There's another reason. If your unloaded Q's are high, you have high circulating currents. Therefore, your coils have to be larger and the current capacity of the capacitor has to be larger.

Because of efficiency requirements and to keep your power bills low, manufacturers have gone to very low loaded Q's. It's not at all strange to find a Q to 5 in an output tank.

In order to get any of these AM stereo signals through, you need to pass the RF wave with the angular modulation through the transmitter. Then you're worried about loaded Q, because that determines the bandwidth for the phase-modulated wave.

We're just very lucky that we don't have to throw out all our transmitters in order to get AM stereo.

All of the exciters will take two stereo inputs, the left and right information. They must pass on to the transmitter a phase-modulated wave which is generally at carrier frequency. Where the transmitter divides them, arrangements must be made to counteract that division.

Generally, however, transmitters are at carrier frequency and there'll be a phase-modulated wave.

In our system, the exciter will provide two or three watts, which will drive a low-level stage in the transmitter. This is done at the crystal stage, actually using the other crystal position which doesn't require any special installation. All you need to do is run a coax lead over to your low-level RF stage.

You also have to feed two audio waves to your transmitter.

That, essentially, is the installation. It is fairly easy, but one thing you don't want to have is a patch that can be reversed.

Once you adjust your system, in order to come out in the proper phase, you want to keep this hard-wired. Your transmitter must be well neutralized, since that creates incidental phase modulation and will cause crosstalk between the L + R and the L − R channels.

After the exciter has been installed and the equipment and cables attached, the adjustment can be done in 20 minutes.

Mr. Payne:

Could you give us an estimate of what the stereo exciter might cost?

Mr. Kahn:

These units are available in quantities of ones and twos at $12,000 a copy. When you get into quantity manufacture the price will be halved.

Mr. Payne:

Arno Meyer will now tell us about monitor concepts.

Mr. Meyer:

As you know, you have two parameters to measure: the envelope and the angle. The envelope is being monitored now and the angle is the additional requirement that must be monitored.

Fortunately, it was suggested that this is all you need to monitor. And I think we'll go along with that because all systems have a required phase deviation they expect to hold.

Now, if they exceed that, then they can get into nonlinearity problems, so there is a necessity to monitor that.

Anything else you may want to monitor or may feel desirable to monitor probably would be your decoded left and right signals. I think the feeling is that it would be up to you.

It would be optional equipment to add on an angle monitor, and there probably will not be stringent specs attached to that, as you have in the FM stereo.

So the monitoring requirement, as we see it, is a simple one and more of a protection-type of monitoring.

Angle deviation would just require an add-on monitor. It has nothing to do with the envelope.

You just sample your carrier, pass it through a phase or quadrature detector, and calibrate that deviation and reflect that reading on a meter having the same ballistic characteristics that you have presently. There probably will be some kind of peak deviation indicator to make sure you don't exceed.

In all of the systems you must pay attention to the negative envelope modulation. Most people like to maintain 100% negative modulation. While these systems won't stand a carrier closure, they'll stand almost a carrier closure. So if you back the modulation down just a little bit — a half db., or something like that — we anticipate no problems.

That's only in the negative direction. In the positive direction, of course, you're free to run it up.

Mr. Payne:

Mike Davis will now discuss audio processing of AM stereo signals.

It's a new area, and I don't believe it will vary from system to system. Generally, you have a difficult problem of maintaining your envelope modulation synchronized with your phase modulation. The limiting characteristics thus are very important.

So, let's hear from a fellow who has worked on the problem in the laboratory.

Mr. Davis:

Thomson-CSF Laboratories, formerly CBS Laboratories, has a fair amount of experience in audio processing. I just thought I'd like to share with you some of the problems in the design of a limiter for use in AM stereo.

Back when we developed the first FM stereo limiter, the question was "How do you do it?" The obvious answer was: "You just take two limiters, couple them together so there's no center image shift." And that was the end of that.

Unfortunately, in AM stereo it's not quite that simple. The sum information, or left-plus-right signal, modulates the carrier in an amplitude fashion. The difference information modulates the carrier in a phase or angle modulation type scheme. The net result is a number of problems.

First, the difference information is angle modulation. Second, the difference information in several of the schemes is pre-emphasized, typically a 100-micro-second curve. Lastly but probably most important, is negative overmodulation or pinching off the carrier which would be highly undesirable.

After a little thought, we came up with an experimental device which is an adaptation of a couple of AM volume control devices.

Very briefly, the device has two paths for the left and the right signals with a variable gain control circuit in both.

A couple of points:

- First, there's no automatic phase-reversing circuit.

- Second, two control voltages are generated. The upper one is formed from the simple sum of the left and right channel; the lower one from the difference information, which is pre-emphasized. Thus two control voltages are generated, the one proportional to the sum and another proportional to the preemphasized difference, or left-minus-right signal. Those two signals are finally tied together, and at any incident time, the greater of the two determines the overall control voltage for both channels.

Since this is an experimental device, there are a number of controls in it which probably would not be present in a final version. For example, there's a pot in the difference channel which can adjust the ratio of control between sum and difference.

We assumed that the stereo generators will like to see at their input the left and the right signal.

I think some advantage can be gained in the limiting process if the sum-and-difference signal can be formed in the limiter. In other words, final safety limiting might be done more appropriately on the sum-and-difference signal rather than on the individual left and right signals.

We've done some preliminary bench tests and found that normally the system performs as expected, in terms of frequency response, distortion, harmonic and intermodulation distortion, and signal-noise ratio.

We're still looking forward to actual program tests, and a couple of these units will be provided to the NAB Engineering group for tests with the various proponents.

One of the questions we hope to answer is, "Will the average modulation be as high in AM stereo as it is in monophonic AM?" My initial feeling is that it will be.

As pointed out earlier, the negative modulation has to be more carefully controlled. We might have to reduce the average modulation a percent or two to prevent any negative stereo overmodulation. But I don't think that will result in any audible decrease in average modulation.

We also hope to find out whether there's 3-db center buildup in this type of scheme, which does not quite occur in the FM stereo modulating scheme, will be a problem for the limiter.

Lastly, will the fact that the difference channel being pre-emphasized, cause an excessive amount of humping or objectionable variation in gain with program material?

Mr. Payne:

What we've said is that you've got to watch your negative modulation. If you go to zero carrier, you have turned off the left-minus-right channel; in other words, you've just turned off the FM part of the transmission when you go to zero carrier and you can't do that.

In order to maintain a compatible left-plus-right signal, compatible for the regular standard AM receivers and be as loud as possible, audio processing problems are extremely important.

I think in the evolution of these things, you might see some limiting in the exciter or some brains in it with regard to control of left-minus-right and left-plus-right modulation.

Let's move on. We've asked Al Kelsch from Magnavox to talk about what an AM stereo receiver might possibly be like.

Mr. Kelsch:

As a receiver manufacturer, we welcome this opportunity to engage in a dialogue with the broadcasters.

My remarks will survey receiver concerns and intentions in three areas.

First, what impact will AM stereo have on the receiver markets?

Second, what features might be present on an AM stereo receiver?

Third, what is the realistic timetable for the introduction of AM stereo receivers.

Magnavox believes the impact of AM stereo on the receiver markets will be dramatic and extensive. Such enthusiasm is shared by many, but by no means all of our industry.

John Love of BUSINESS WEEK, in a March 21st, 1977 article on the subject of AM stereo, describes the situation in the following way: "The new service (AM stereo) will be a boon to radio manufacturers, increasing their two billion dollar retail market by a possible 400 million dollars or more."

As a manufacturer of stereo equipment, Magnavox has felt from the outset that we dare not ignore the developing AM stereo service. We view the AM stereo picture as a positive opportunity to stem the foreign onslaught, at least within our traditional markets, which is console stereo.

This realization led us first to support the work of the National AM Stereophonic Radio Committee, and finally to our entry into the AM stereo arena as a proponent, which we did on December 14th of last year.

What features will be presented on an AM stereo receiver? It is our belief that many of the features present on the front panel of an FM stereo receiver will eventually be found on an AM stereo receiver.

First and foremost, we believe, a stereo indicator will, be necessary for full public acceptance of AM stereo. Since our business is the marketing of receiving equipment, we feel this feature cannot be overemphasized as a necessary part of the final AM stereo product.

The stereo light is a feature which can be demonstrated on the showroom floor and one which the public has come to expect when dealing with a stereo system.

Interstation mute capability will be offered on all but perhaps the most basic AM stereo receiver. This again follows the FM stereo lead.

Since the stereo receiver, in most cases, will use limiting in some portion of the signal processing, off-station noise will be high and mute will be required.

A front-panel selectable dual-bandwidth feature will be offered.

If the AM sound is to be in any meaningful sense competitive with FM, bandwidths out to perhaps plus-or-minus 12 kHz will be realistic.

For quality reception in high-signal areas, perhaps in the 25-millivolt-per-meter category, the consumer will have a wide bandwidth available. In lower field strength areas, the narrow bandwidth position, comparable to the bandwidths existing on current AM receivers, will be useful, since the wider bandwidth position would also mean additional background noise in weaker signal areas.

In this context, I take this opportunity to invite the industry to begin to plan for better-quality AM.

I realize that the AM broadcasting industry has traditionally processed the audio signal in the expectation that narrow bandwidth receivers were all that was available, and hence full dynamic range transmission was a waste of modulation capacity, and hence loudness.

To fully utilize the capability of the new receiving equipment, a full dynamic range transmission will perhaps be an attractive broadcaster option, in the same sense that such an audio signal is attractive to the FM broadcaster.

In the new era of AM stereo, a program product which is the equal of FM in every important aspect will be possible if we, the industry, elect to provide it.

The third question: "What is a reasonable timetable for the introduction of a new AM stereo receiver after adoption of a system by FCC?" It is my opinion that the domestic receiver industry will make equipment available very promptly after a Rule Making.

If the Magnavox system were to be selected, this date could be as short a period as six months.

In summary, while the entire receiver manufacturing industry may not react with uniform enthusiasm and purpose, I do assure you that at least one receiver manufacturer will respond to the AM stereo decision with a sense of optimism and opportunity. To accomplish this, a line of AM stereo receivers will be offered which will be capable of utilizing the full capacity of the new service and will compare favorably with FM equipment, both with regard to bandwidth and front-panel features.

Mr. Payne:

The last panelists is Harold Kassens, who is going to describe the activities of the National AM Stereophonic Committee and other efforts in behalf the AM stereo before the FCC.

Mr. Kassens:

In September, 1975, the NAB, the IEEE Group on Broadcasting, the National Radio Broadcasters Association, and the Electronic Industries Association joined together to sponsor the National AM Stereophonic Radio Committee. We notified the FCC that the Committee had been formed for the purpose of 1) examining in detail all the systems which were presented to it; 2) to test these systems and 3) submit the results to the FCC for a decision.

The Committee has organized and has been hard at work. There is a steering committee which supervises the entire operation and four supporting panels:

Panel One — Systems Specifications is headed by Carl Eilers of Zenith. It's purpose is to analyze theoretically all the systems and give guidance and direction to the other panels.

Panel Two — Transmitter Characteristics is chaired by Granville Klink of WTOP. It's purpose is to examine the transmission systems as they relate to each one of the proposed systems to see how the system will react under various conditions.

Panel Three — Receiving Systems is headed by Tom Pruitt of Delco Radio. It's purpose is to examine each systems, from receiving antenna input to loudspeaker output, and to determine performance.

Panel Four — Field Tests is headed by George Bartlett of NAB. It's purpose is to field-test each of the systems and to assist in the preparation of the final report.

At the beginning, we had proposed to us four systems: RCA, Sansui (with two systems) and Comm Associates of New York. As we were analyzing each one of these systems, all proponents withdrew. However, as those systems were being withdrawn, the three systems we now have under consideration — Motorola, Magnavox and Belar — were proposed to the Committee and have now been completely analyzed.

We are not, in our Committee, considering the Kahn system in any way, shape or form. Mr. Kahn has his own system and has chosen to go his own way with the FCC, which, of course, he's entitled to do.

A final field test plan has been drawn up. We have selected three stations to join in the field test, WTOP Washington, D.C., 50KW, 1500 kHz for ground wave tests; WGMS Washington, D.C., 5KW, 570 kHz for ground wave tests; and WBZ Boston, to test skywave propagation.

We will start out with laboratory tests under controlled conditions, where we can determine system performance in to an idealized receiver, or a monitor; and, secondly, under normal conditions with a typical receiver. We will measure frequency response, distortion and separation. We will also determine mono-compatibility.

We have selected types of receivers to determine the effect of each one of the systems on mono-compatibility.

We also will determine receiver audio power, stereo and mono sensitivity; occupied band-width, and protection ratio. One of the important things, as far as the FCC is going to be concerned, is what does this do to interference?

We will test these parameters under laboratory conditions, and then move into the on-the-air tests, where we will also determine system performance over-the-air.

We're going to make mobile measurements to determine the extent of distortion in the nulls of directional patterns. We are going to use WBZ to determine the effect of selective fading in skyward propagation on stereo signals.

So far, we have run into a few interesting problems such as incidental phase of transmitters. We are now going to have to worry about what the incidental phase, or phase differentiation, of AM transmitters is.

Fortunately, the three stations we have selected have three different transmitters, and we will be able to get some information.

This matter has also been called to the attention of the transmitter manufacturers in the hope that they will start worrying about the problem.

Another problem is the Q of the circuit. We now have to worry about the antenna impedance, particularly the commonpoint impedance of a directional antenna. The impedance curve in amplitude and phase is not linear.

What is the effect of this complex impedance on a signal which is now being modulated both in amplitude and phase? We've done considerable work in the transmission systems panel on the effect of varying impedance, and, hopefully, our tests will tell us the degree to which the licensee is going to have to worry about antenna impedance.

It's been amply demonstrated this morning that modulation is going to be a problem. What happens when you modulate an AM stereo signal 125% positive or 124.9% positive and 100%, on the dot, negative?

We have notified the FCC that we intend to begin the field tests on May 2nd here in Washington. At most, the tests are expected to take two months.

We have also notified the Commission that we will submit the report of our complete work to them in late summer, before Labor Day. At that time the FCC will issue a Notice of Proposed Rule Making. Time will be allowed for the submission of Comments and Reply Comments.

Subsequently, at some future date, a Report and Order selecting one system will be adopted.

I would guess it would take the Commission at least six months to arrive at a decision after they get our report. Maybe, if we can put enough pressure on them, we'll have their answer by the next NAB Convention.

Anybody is welcome to come to any of the meetings. I am urgently recommending that all manufacturing people have a representative at our meetings.

Mr. Payne:

I think it would be fair if we let Leonard Kahn explain why he is taking the route he is.

Mr. Kahn:

I've heard a number of times that in AM stereo, you have to avoid 100% negative modulation. Completely untrue.

We have 3½ years of experience at XTRA and didn't touch the negative modulation one bit.

There is no reason for you to give up fully modulating your transmitter. If you want 125% modulation, fine. If you want to do 100% negative or slightly less that's also fine.

There is no breakup with our system.

I think you're going to find that you're going to have the same breakup in AM stereo as you have in FM stereo if you do anything but utilize the independent sideband approach. I'm very earnest about that. This is simple analysis, and it's about time that the committee faced up to that problem.

The second thing is the need for an indication of AM stereo presence. We agree fully that it should be there.

We have a number of manufacturers, unfortunately, most of them outside the country, who will make sets. You don't have to wait six months. The day you go on the air, you can have your listeners hearing stereo using two receivers. The inexpensive special sets can then be available six months later, from all the manufacturers.

Mr. Payne:

Is there anyone else that would like to comment in any way, either on what Mr. Kahn or anyone else said?

Mr. Kassens:

Leonard is welcome to his system. If he wants it tested, we'd be glad to test it. We don't think he's tested his system and he's entitled to his opinion.

Mr. Payne:

Are there any questions?

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Question:

What would be the effect, if any, on fringe coverage with AM stereo as compared with mono AM?

Mr. Parker:

Generally speaking, there shouldn't be a great deal of effect on the fringe.

I would guess, for instance, that in a system here you had 150-miles ground-wave-coverage, that you might reduce that range, at most, 20 or 30 miles. In the outer fringe of that range monophonic reception would still be available. 20 miles in, you would get stereo with the same signal-noise ratio as essentially the outer range.

Let me say that the systems will vary somewhat differently in the presence of amplitude modulation. That is, the background noise in an unmodulated carrier will be essentially as I have described it. There might be a difference in the behavior in the system in the presence of heavy modulation.

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Question:

Have you checked a common point that is far from linear? One that would appear to be 50 ohms on frequency and maybe 70 ohms out about 8 KC on one side and down to 25 or 30 ohms on the other side?

Mr. Kahn:

Yes. This is a very important question. Independent sideband, which is essentially CSSB, is an independent single sideband system. You can upset the phase and the amplitude of the sidebands when they're individually set.

One of the main advantages of single sideband is that the relative ratio and phase of the sidebands isn't important, as it is to regular double sideband where they must be perfectly symmetrical if you're going to avoid distortion.

WBFR has a fairly directional array and XTRA has a five-stick array that is extremely sharp. We had absolutely no problem.

I think you're going to find that most of the systems aren't too bad on this. I think we're worrying too much about it.

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Question:

In a common-point impedance, do you feel this system you're going to test is fairly flat?

Mr. Kassens:

Well, as of now, we have checked one of the three. It is not fairly flat.

The problem is that what we're really talking about here is the impedance at the plates of the final stage, not at the common point. This is where you're really generating the signal.

My comment would be: If Leonard Kahn says he didn't have any problem, I don't know how he knows unless he measured it, unless he rotated the phase of the transmitters. I rather doubt that he went into the transmitters to work on this.

And if you're talking about listening over the air, I say, "How good are your ears?"

I think it's a concern. We have a gadget we're now building up in our office to vary Q's from roughly about 5 to 30, which we're going to use in the test to determine how big the magnitude is. It's a problem. We owe the FCC an answer, and we're going to give it to them.

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Question:

My first questions relates to Class 1B stations. What is the Commission's position going to be about Class 1B stations that make an arrangement with another Class 1B station that "You get the upper side band and I get the lower," or "You get left and I get right?" for nighttime coverage?

Mr. Payne:

That's been tried between WBZ and KDKA, and I think it was Mr. Kahn's equipment that did it. And it wasn't continued. But we understand your point.

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Questioner:

The second question is also relative to selective fading. What measurements are going to be made of stereo performance under selective-fading situations?

Mr. Payne:

The tests are going to be an intermodulated test. We're going to transmit two tones on WBZ for about a half an hour. The intermodulation will be recorded as the station goes through selective fading.

You go through four different points of a matrix. You'll be transmitting AM and then stereo, and you'll have two receivers on the receivers end, an AM receiver and an AM stereo receiver.

So you'll have all four conditions: 1) AM on an AM receiver; 2) AM on a stereo receiver; 3) stereo on an AM receiver, and 4) stereo on a stereo receiver.

You'll be measuring the intermodulation distortion as you go through fading. You'll take the median value of the intermodulation distortion over the period of time and try and come up with a factor of the intermodulation distortion difference over a period of time.

That appears to those on the committee to be a fairly decent way to do it.

Mr. Parker:

I think you're going to find in selective-fading measurements that for most of the night, the signals are solid; and that when you can hear the signals clearly in mono, you will get them clearly in stereo. At times when they're distorted in mono, they will be distorted in stereo.

In other effects, when you get the barrel effect, when the sidebands are rotated, that will give you a problem with stereo. But when you hear it clearly, as you do most of the night, you'll find that the stereo will work equally well.

Mr. Kahn:

We ran a lot of skywave experiments on WFBR and XTRA.

I really don't think you're going to find a tremendous difference between the systems in terms of this problem. However, I don't think you are going to get anywhere near skywave on the other proposals because they're going to break up on low signal-noise ratios.

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Question:

Leonard, with regard to items such as limited-bandwidth antenna systems and skywave, have you done any measurements, and can you tell us what you found?

Mr. Kahn:

The AM broadcasters point of all this is, "How does it sound in the field?"

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Question:

How about the fading area in the intermittent service area of a station? What tests are being made as to the suitability of reception in stereo during the intermittent service area?

Mr. Payne:

That's another form of selective fading.

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Question:

In regard to compatibility of the various proposed stereo systems and in reference to the existing mono receivers in the field, particularly with regard to perceived frequency balance of receivers, there's a certain fiction that AM is broadcast without pre-emphasis. But, as we all know, because of industry practice, mainly because of extreme narrow banding of conventional mono receivers, this is no longer true.

I heard at least one proponent say that he was proposing a 100-microsecond pre-emphasis in his difference channel and a presumably flat response in the L+R channel.

I would propose that if these systems are indeed broadcast flat, that they would be almost unlistenable on a typical mono radio. And therefore I'm wondering what people propose to do about the problem of spectral-balance compatibility between the so-called high-fidelity stereo radios and the millions of existing mono radios in the field.

Mr. Davis:

I can't see that would be any particular problem. Many AM stations are now inserting some type of permenent or transient variable-presence boost. I'm not certain how it would make any difference in terms of compatibility.

Mr. Payne:

Don't you think that if people wanted to tweak up their frequency response, they could still do it. And the technology's a little different, because of the way AM stereo would work, but don't you think it would still be their option?

Mr. Davis:

What particularly disturbs me is any proposal which puts a different pre-emphasis on a difference channel than on a sum channel. I worked for two years on the problem of getting decent-sounding, relatively flat sound out of narrow-band receivers.

But if you put another 100-microsecond pre-emphasis on top of that, the situation becomes totally out of hand. And I think any system that's going to use pre-emphasis must have identical pre-emphasis on the sum/difference channels.

Mr Payne:

Because of processing problems?

Mr. Davis:

Yes, because of power-spectral distribution problems with real program material.

Mr. Kelsch:

Our notion of providing a dual-bandwidth receiver is probably relative to the question. I think you're suggesting on a full-fidelity wide-bandwidth receiver that the signal you're talking about will not sound good on program material.

I agree, I think that's why new stereo equipment will have the option. If you don't like what you hear, you switch to the other position. My feeling is that there should be a standard AM pre-emphasis curve.

Mr. Payne:

That's about all the time we have.

Let me thank the Panel for their cooperation and understanding and your kind attention. We will continue in the Tudor Room with a demonstration of the AM stereo equipment.

Beyond Electronic News Gathering (ENG) Workshop

Plus, Questions and Answers

Presiding:

Frank L. Flemming
Vice President, Engineering
NBC Television Network
New York, N.Y.

Moderator:

K. Blair Benson
Telectronics International Inc.
New York, N.Y.

Participants:

Mr. Flemming

Joseph A. Flaherty
Vice President, Engineering and Development
CBS Television Network
New York, N.Y.

Isaac Hersley
Equipment Planning Engineer
American Broadcasting Company
New York, N.Y.

Richard T. Monroe
Vice President for Engineering
Westinghouse Broadcasting Company
New York, N.Y.

Merle Thomas
Associate Director for Technical Operations
Public Broadcasting Service
Washington, D.C.

Mr. Flemming:

On behalf of the Society of Motion Picture and Television Engineers (SMPTE), welcome to the workshop on "Beyond Electronic News Gathering." We are pleased to present the workshop for you.

It may not have been too clear in the program what we're going to do today, so let me explain. We have two short papers, both supported by videotape demonstrations, about cameras and low-cost one-inch video tape machines. Following the papers, we'll have a panel of four experts, who will answer any questions you may pose.

It's my pleasure at this time to turn this session over to your moderator, Mr. Blair Benson.

Mr. Benson:

Our first presentations this morning will be by Joseph Flaherty, Vice President of Engineering and Development for the CBS Television Network and past Vice President of Television Affairs for the SMPTE.

So, with no further words, I'd like to introduce Joe Flaherty.

Mr. Flaherty:

Thank you Blair, and welcome.

It was at this conference just two years ago that Julius Barnathan, ABC, New York, N.Y., said on the ENG panel, that those of you who could should, "get your toes wet and try the ENG water."

Last year, at this same conference, he said, "It is time to wade in, but to try not to drown."

Today, there are more people in the pool than sitting on the edge. But here on the floor the other day, I met one of the edge-sitters who couldn't understand what all the splashing was about.

It reminded me of the story that came out of Spain when Generalisimo Franco was dying. Laying on his death bed, surrounded by his friends and advisers, he heard from the outside the roar of the crowd. Turning to his closest friend, the General said, "What's all the noise?"

His friend replied, "There's thousands and thousands of people out there, General."

And he said, "Why have they come?"

And he said to him, "Well, sir, they have come to say goodbye to you."

Whereupon the general asked: "Really? Where are they going?"

What lies beyond ENG? We expect an expanding horizon for electronic photography at both the local and national levels.

At the local level we will see expanded use of equipment that's now in everyday service in the production of local documentary programming, community programming, and local retail commercials — fields which have been difficult if not impossible for most stations to cope with heretofore.

At the national level we'll see the use of both ENG equipment and the more sophisticated one-inch videotape machines in the production of national documentaries, the increased production and distribution of national commercials, and in primetime entertainment programming.

I think the best way to describe the local documentaries is to show you a mini-documentary which was shot on location with a single camera at night. It was produced in Chicago by WBBM-TV and is entitled "a Taxi-driver's Chicago."

(Tape rolls.)

I'd like to show you next another documentary, shot in Alaska, covering construction of the Alaskan pipeline.

The CBS News Bureau in Los Angeles took the ENG equipment to Alaska. The scenes were shot in subzero temperatures with a wind-chill factor of minus-40 degrees Fahrenheit. The helicopter shots were made by hand-holding the Ikegami HL-35 camera out the doorway, where the outside temperature was minus-40 degrees.

The only malfunction was the automatic lens iris, which had to be operated manually.

The crew recorded a total of 28 video cassettes and lost only three minutes due to the cold. The tape recorder was protected from the cold and moisture by being kept in a Kelty backpack as it was carried over the tundra for some extended shooting sequences.

(Tape rolls.)

So much for documentaries. We could go on and on with this sort of thing. It's a harbinger of things to come. A greater utilization for ENG equipment, and provides a service theretofore impossible on a practical basis by most stations.

To help set the stage for our roundtable discussion, I'd like to move onto the subject of one-inch tape. I'd like to show you some of the comparative results we've achieved in some early testing on the Sony BVH-1000 version of the one-inch videotape machine.

For the first demonstration, I'd like to show you a piece of tape that was recorded live from a Television City show and subsequently was dubbed down to the 12th generation.

For this demonstration, I've had to transfer from the original so you'll see a short take from the 2nd generation, and then the 12th generation of the same material, picture and sound.

This sort of flexibility, coupled with the small size and low cost of such equipment, prompts us to press it into service for primetime entertainment programs as quickly as possible.

CBS is planning an installation this spring at our film studios in Hollywood, to produce some of the programs heretofore produced on tape.

To prepare for that, we produced another test which compared 35-mm film with a Thomson 1515 studio camera feeding the Sony BVH-1000, and the Thomson microcam, the little eight-pound ENG camera, feeding the same high quality BVH-1000 Sony tape recorder.

While most of you aren't really interested in 35-mm film, it was important for the production. The test begins with a short piece of an air-copy tape of The Bob Newhart program without sound. We did that as a benchmark against which we could measure the quality of the 35-mm test.

First you will see a short piece of Bob Newhart. Then you see the beginning of the test with the 35 mm film. This is followed by the Thomson 1515 camera and the Sony BVH-1000 and the microcam, Thomson microcam with the BVH-1000.

All of this was subsequently film-transferred to two-inch tape in the normal way for broadcast. The helical-scan tapes were transferred to quadruplex and edited.

Today, it's all been put on Sony BVH-1000 so we can demonstrate it here with a Thomson studio camera, the microcam, and the 35-mm Mitchel.

(Tape Rolls)

First, the Bob Newhart piece for calibration purposes.

Next, our 35-mm millimeter film transferred to tape.

Then the Thomson 1515 studio camera on the Sony BVH-100.

Next is the little eight-pound microcam feeding the Sony BVH-100. Then to the 35-mm film.

The Thomson studio camera and then the microcam.

Moving now to the backlot and the 35-mm film.

You should recognize that street—four hundred million people have been shot there.

We show it with the Thomson studio camera and the Sony BVH-100. Then the microcam with the Sony-1000. The film again, and finally with the Thomson studio camera.

Mr. Benson:

Thank you very much, Joe, for a very interesting and exciting discourse on our new developments. Any questions from the audience?

Question:

I'm just curious. Did you use the normal film/tape transfer process and techniques?

Mr. Flaherty:

That test was done by the normal film procedure in Hollywood. The normal CBS procedure is that all the film shows are transferred to videotape in Hollywood for broadcast from New York.

This test was done the same way by the same people on the same chains normally used. It was not meant to be a scientific test; it was an operational test.

Question:

Was there any digital noise-reduction used?

Mr. Flaherty:

There was no noise-reduction used during any of the recordings. However, the first two ENG-style documentaries, which were 5th generation, were run through a noise-reducer. The others were not.

Mr. Benson:

Our next presentation is by **Isaac Hersley**, equipment planning engineer, ABC, New York, who will give us another insight on "Beyond ENG."

Mr. Hersley:

What I'd like to discuss this morning is somewhat different from what Joe just spoke about. I'm going to emphasize the increased reliability of ENG in the past year or two and the use of such equipment in the sports and entertainment areas.

Basically, one-camera pickups consisting of an ENG camera linked to the studio by microwave were often utilized during the 1976 election year. In years past, large studio-type cameras were required but now an ENG unit, with its microwave system, could be assigned to the remote set up, quickly set up and ready for the telecast, then again broken down as quickly as the set up.

One camera, the HL-33, was fitted with a four-inch viewfinder and a gen lock was installed in the camera backpack when a second camera was added.

Adding one more camera module to the system described—namely, the base station—permits the engineer to mate an ENG camera with a full-sized production mobile unit or supplement studio cameras.

With triax cable, the camera could be a maximum of 1600 meters from the base station. Complete controls for camera setup and iris controls permitted the video operator to match this camera to the other cameras in the mobile unit or the studio.

Coax was also used, but for power the camera had to run off a battery or a local power supply.

For the Democratic and Republican conventions, two mobile units of the motorhome type, each with three Ikegami HL-33 cameras in base station, were positioned for coverage at the hotels of the presidential nominee hopefuls.

The use of this size camera and its quick connect/disconnect coaxial hookup permitted connection to as many as 12 pre-wired drops throughout the hotel complexes.

Coax and not the more expensive triax was used. Coax is much more maneuverable through shaftways and ducts, taped down to floor and carpeting and over the rooftops of the hotel. In addition, the unused coax lines were capable of accepting an ENG crew with backpack only when additional cameras were required for coverage of a particular event.

In 1974, ABC commissioned Ikegami to design and produce an RF camera system based on the HL-33/35 camera.

The result was an elaborate modular system which simply required an additional box mounted on top of the backpack. In addition, receiver antennas in base stations were also designed and supplied for this RF system.

The signals are transmitted to the four-foot dishes located at various points in the coverage area.

Three backpack transmit antennas are available for use, a 20-degree, a 60-degree, and an omni-directional. The first permits approximately one mile from camera to receiving point, while the second and third types provide three-quarter and quarter-mile ranges. The backpack gains approximately nine pounds in this RF mode.

The backpack transmits with a 13-gigahertz carrier, while receiving a 950 megahertz telecommand signal from the base stations.

Additional items at the base station required for the RF camera include a decoder and an NTSC encoder.

Again, a basic ENG item was utilized. The camera had been set up for RF operation.

One of the first uses for slant-track tape, for other than ENG uses, was for the protection of daytime ABC-TV programs. Quadruplex equipment had backed up original program VTRs at considerable cost.

Facilities were provided so that two Sony 2850s and time-base correctors were dedicated to daytime backup of the two-inch program material.

The mini-mobile units, which were mentioned earlier, did not carry two-inch tape equipment at the political conventions. Instead, all units were equipped with slant-track recorders.

In Kansas City, all ABC units assigned to locations from the convention hall contained two Sony 2850s and RM-400. Material was recorded when necessary and played back directly to air when called for by the director back at the convention hall.

Probably the most unusual utilization of the 2850 was at the Winter Olympics in Innsbruck.

To set the mood for the games, helicopter shots of the magnificent scenery were recorded. This material was captured using an HL-35 camera and a 2850. Immediate playback of the recorded tape through camera viewfinder permitted quality checks of the footgage.

Interestingly, when the 2850 was stored in heated rooms overnight and then brought out to the helicopter the next morning, moisture would prevent immediate recording operations. With this in mind, ABC personnel left the 2850 units in sheltered, unheated buildings overnight, and then the next day found they had no problem. They were cold but they worked very well.

Portable recorders are part of the ENG news operation.

Electronic Sports Gathering, (ESG), is a new name for the same thing by a different department at ABC. An ESG crew covered the headquarter hotel for the Monday night baseball series last spring and summer. Likewise, the concept also was soon used for NCAA football, NFL Monday night football, and both Winter and Summer Olympic Games.

Time-base correctors are an integral part of the cassette playback-to-air operation.

The ABC news ENG crews carry compact units for playback in the field. Still other units are required in the large-scale mobile units for playback of ESG.

The TBCs in the mobile units and those at various ABC studio facilities also function as playback-to-air devices for Arvin/Echo framestore recorders.

What's ahead in ENG?

We've all seen vast improvements in cameras. Smaller ones, better ones, lighter ones, and cameras that consume less power.

As for tape machines, we still have the three-quarter-inch cassette. A half-inch tape scheme of quality equal to 16-millimeter tape, with handling characteristics of the present cassette, is the next logical step.

A portable recorder of this type must be small, lightweight, and consume as little power as necessary, so that the operators do not have to change batteries too often and can keep up with the fast pace of the news crews.

I want to show you some tape samples of various news pieces and some shots from the Olympics and then you can see how ABC uses this equipment in areas other than ENG.

(Tape rolls)

Mr. Benson:

Thank you very much, Isaac.

Our next item on the program is a panel discussion during which we'd like to stress the new one-inch equipment. Joining us on the panel will be **Joe Flaherty**, our first speaker, **Frank Flemming,** Vice President of Engineering for NBC; **LaVern Pointer,** Vice President of Broadcast Engineering for ABC; **Richard Monroe,** Vice President of Engineering Westinghouse Broadcasting Company; and **Merle Thomas,** Associate Director for Technical Operations for PBS.

As we all know, the introduction of these new broadcast-quality one-inch formats have posed a serious problem to all broadcasters and production houses in the matter of interchangeability.

The problem, in fact, is significantly more serious than that which was encountered with the introduction of high band and super high band quad equipment. The reason is that the basic recording format is the same for all of the various quad types of recordings, but the one-inch formats are entirely different and are not interchangeable.

The question is: "How do we plan to introduce this type of format to the broadcasting industry, and what steps are being taken to bring about some form of interchangeability?"

I'd like to ask Merle Thomas, to bring us up-to-date on what industry activities are underway in this field.

Mr. Thomas:

At the winter SMPTE television conference, CBS and ABC proposed the standardization of one-inch machines. Two working groups were set up, one for segmented scan and one for non-segmented scan. Fred Remley is the chairman of the non-segmented working group and I'm the chairman of the segmented working group.

Commercial users, all networks in the United States, as well as CBS in Canada, are represented on both of these working groups. Each group has had two meetings, and at this point discussions are taking place as to what the standard should be.

The segmented working group has received a standardization format proposal. The meetings will continue until a standard is developed.

Mr. Benson:

Joe Flaherty in his presentation, mentioned what CBS is doing in Hollywood.

Joe, could you expand on how you plan to cope with the various types of one-inch, the quads, super high band with pilot tone, three-quarter-inch, umatic, and so on?

Mr. Flaherty:

Actually, the question of standards is a critical one in any field. We've faced this many times.

The danger is that one can standardize so soon as to cut off innovation, to cut off further development, and cut off the natural development of new technologies.

On the other hand, one can wait so long that there is such a proliferation of proposals that there is chaos in the market.

We, hopefully, are somewhere between those two points right now.

What makes a standard happen, I think is one way to look at that. There's a combination of technical advantages: improved quality or reliability. There also are physical advantages: simplicity, improved flexibility. And there are physical advantages: small size, low cost, economic advantages, and so on.

These can combine in a myriad of combinations to effect a need for a new standard. But there is a plateau; there's a threshold level.

In one-inch tape, I think we're all above the threshold level. The machines are a third of the size of quadruplex machines. They're a half to a third of the cost. They have about a third of the operating cost, in terms of heads and tape and tape size and tape storage.

Unless they fall apart between here and there, they're going to make it.

We at CBS hope that we've bought our last quadruplex recorder. We intend to use this new format as much as possible to take advantage of these technical, operational, and financial advantages. And I think that's the way I'd characterize it.

Mr. Benson:

Perhaps we might get another viewpoint. Westinghouse, I understand, is the largest air freight forwarder in

Pittsburgh because of the large number of tapes that are shipped out every day for syndicated programs.

How do you plan to cope with this, Dick?

Mr. Monroe:

I'm looking for a one-inch tape format that we don't have to worry about what machine we put it on. We want to send out a tape to a broadcaster and he can put it on his machine, no matter who manufactures it, and play it. That's how we feel about it.

Mr. Benson:

Anyone have any questions they'd like to pose to our panel?

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Question:

Could you comment on the Umatic concept?

Mr. Flemming:

Umatic is a great format for hard news but there is a riding speed limitation.

We don't see it becoming a high-quality machine, like quadruplex is now, and as one-inch certainly appears to be. We see continuation of two separate uses.

Mr. Benson:

We might add that the three-quarter inch is a color-under recording system, that really doesn't provide the full quality that is acceptable to the broadcaster. It doesn't provide the full MTSC bandwidth and quality.

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Question:

My principal interest at the moment is the effect all this new technology has on people.

The technical people who were formerly necessary to operate quads in order to get any sort of picture at all are not necessary to operate this newer equipment. Camera technicians are no longer required, because creative people pushing buttons while looking at white cards take care of all kinds of automatics, even including registration.

The question is what plans are you, as users, trying to develop to retrain the present or to find people with the new skills which are required?

Mr. Pointer:

I think you have to understand that although technology is improving, our labor costs are continuing to rise. We have to offset the costs in terms of technology, operational requirements, manpower, and so forth.

We basically look at equipment today and say: "What will this give us in terms of an economical offset on rising labor costs?"

The question is not whether to replace X number of operators, but how can we use them better, and perhaps enable them to do their job a little better.

In the area of videotape, the editing cost is one of the primary areas that is getting completely out of hand. The amount of editing time must be brought back to manageable levels, not only for the individuals home life but also in terms of the operating costs.

We look to one-inch videotape as an area which is going to improve our editing and hopefully reduce our editing costs which are getting completely out of hand.

Questioner:

Improve the editing efficiency over quad?

Another Panel Member:

One of the ways that the one-inch format got started was by specifications for editing equipment.

We are moving more and more into the film style of photography to produce programs that have up to now been produced largely on film—that is to say, the single-camera technique, as we did in ENG. We solved the single-camera technique for ENG. We now have to solve it for first-class production.

As we move in this direction, we must have the same kind of editing flexibility that you have in film; namely you have to be able to study the edit points, you're not going to make them on a switcher in that case. The so-called production edits will disappear in one-camera photography. They'll be made in post-production.

So you'll have to be able to study the edit points in still frame and in slow motion, and you have to be able to have the flexibility that film has had these many years, with horizontal editing tables and movieolas.

This is where the big advantage of the helical formats have over the quadruplex.

Another Panel Member:

One important point is that because of the high capital cost of equipment, it's been necessary to utilize the equipment practically around the clock.

With less expensive machines, this capital equipment investment is reduced and the need to run the equipment around the clock is also reduced to some degree.

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Question:

What kind of attitudes do you find reflected among the unions? Will they be willing to give up some of the traditional types of jobs?

Mr. Benson:

Can you be a little more explicit?

Questioner:

My general feeling about union negotiations in the past has been that: "We're very happy to cooperate with you, provided you maintain the rights that we have to equipment."

This is particularly pertinent in the camera area, where technicians have been *the* camera operator, and are no longer necessary *the* camera operator.

Mr. Benson:

Okay, Gentlemen, we thank you for your contribution. A very interesting program.

Notes