NATIONAL CONVENTION

BROADCAST ENGINEERING CONFERENCE

Proceedings of the 1987 SBE & Broadcast Engineering Conference

Journal of the Society of Broadcast Engineers

1987 SBE National Convention and
Broadcast Engineering Conference

A. J. Cervantes Convention Center
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November 10, 1987

Proceedings

One of the biggest challenges facing a broadcast engineer today is how to stay current with rapidly changing technology. While the benefits afforded by new technology may be great, so too are the responsibilities placed on those who must operate and maintain such hardware. The technical demands facing a broadcast engineer or operator today are greater than ever before. However, your attendance at this conference affirms your desire to meet the challenge and learn about many of these new developments.

Convention schedule

Tuesday, Nov. 10: *joint radio/TV sessions*

10 a.m.	Welcome: *John Battison, conference chairman*
10:20 a.m.	Master-clock and time-keeping systems for broadcasters
10:50 a.m.	Improved grounding methods for broadcasters
11:20 a.m.	Fundamentals of digital audio
11:50 a.m.	Where is broadcasting going?
12:30 p.m	Lunch
1:30 p.m.	Maintenance seminar
7:30 p.m.	The nuts and bolts of audio processing

Wednesday, Nov. 11: *radio sessions*

7:30 a.m	Surge protection and grounding for AM transmitter sites
8 a.m.	Telephone hybrids—what they can and cannot do
8:30 a.m.	The impact of 75μs pre-emphasis
9 a.m.	The role of the audio switcher in radio
9:30 a.m.	Automation in the application of direct-to-air CDs
10 a.m.	Coffee break
10:15 a.m.	Getting the most from the NAB cartridge
	system
10:45 a.m.	Theoretical development of the folded unipole
11:30 a.m.	Lunch
12:30 p.m.	Electronic broadbanding of AM antennas
1 p.m.	Fundamentals of digital audio
1:30 p.m.	Automatic phase-correction for tape-cartridge machines
2 p.m.	Measuring soil conductivity inexpensively
2:30 p.m.	Dial telephone remote control
3 p.m.	Coffee break
3:15 p.m.	Report on new reduced-skywave antenna tests
3:50 p.m.	NRSC progress report
4:30 p.m.	Consultants' round table
5:30 p.m.	The FCC round table (*joint radio/TV session*)
7:30 p.m.	Nuts-and-bolts session (*joint radio/TV session*)

Wednesday, Nov. 11: *TV sessions*

7:30 a.m.	True APL picture of power of a TV transmitter
8 a.m.	Still-image library management
8:30 a.m.	Quick TV stereo proofs
9 a.m.	Uses for the network analyzer
9:45 a.m.	Correction of phase-delay error
10:15 a.m.	UHF multichannel TV antenna systems
10:45 a.m.	Coffee break
11 a.m.	A new era in video measurements
11:30 a.m.	Trends in TV audio; and Monitoring the BTSC signal
12:30 p.m.	Lunch
1:30 p.m.	Mobile mast safety considerations
2 p.m.	The engineer/artist, interface in computer graphics
2:30 p.m.	Fiber optics—the next medium
3 p.m.	Coffee break
3:15 p.m.	Advanced TV service: the next generation of TV service
4 p.m.	The 15kW Klystrode
4:30 p.m.	Consultants' round table
5:30 p.m.	FCC round table (*radio/TV joint session*)
7:30 p.m.	Nuts-and-bolts session (*joint radio/TV session*)

Thursday, Nov. 12: *joint radio/TV sessions*

7:30 a.m.	Frequency coordination and how to set up a frequency coordination committee
12:30 p.m.	Luncheon: *Wally Johnson, speaker*
2 p.m.	The engineer in transition (a panel discussion)
4 p.m.	Goodbye until next year: *John Battison, conference chairman*

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Proceedings

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Proceedings Note:

All authors providing manuscripts for publication in these **Proceedings** were requested to keep their submissions generic in nature and not related to a specific product. Authors were further asked to avoid using company names unless necessary for completeness and accuracy.

Because of the time and staffing limitations involved in the publication of a large volume such as this **Proceedings,** no attempt could be made to edit manuscripts to conform with the guidelines set forth when the work was commissioned.

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TRENDS IN TELEVISION AUDIO

Randall Hoffner
National Broadcasting Company

Abstract

The electronic sophistication of the American public, the competitive nature of the video marketplace, and recent technological innovations have generated and sustained a keen interest in high-quality stereophonic television audio. We have come a long way in a few short years, and ongoing developments are creating an exciting future for the sound portion of the aural and visual medium of television.

For much of United States television broadcasting's history, the sound that accompanied the visual image was accorded the status of "second class citizen" and, indeed, sometimes appeared to be little more than an afterthought. Our system of transmitting television aural signals employs frequency modulation and has thus always contained the potential for high-fidelity sound transmission, and there was ample bandwidth available for the inclusion of subcarriers to support multichannel sound. That television audio failed to realize its potential for so long may be attributed to a number of causes. Terrestrial network delivery systems compromised the quality of audio delivered to affiliates. Video tape recorders were optimized for video performance to the neglect, if not the outright detriment, of their audio performance. Studios and sound stages were often acoustically deficient and noisy.

Television receivers had "low-fi" sound sections consisting of anemic power amplifiers driving small, low-quality speakers. These receivers delivered unexciting sound to the viewer, but did offer the benefit of filtering out the buzz created by such transmission impairments as incidental carrier phase modulation and visual carrier over-modulation.

These factors, coupled with a general lack of interest by the industry in fully exploiting the potential of television audio, caused the sound portion of the aural/visual medium of television to languish for several decades. Through the years there had been some inquiries regarding stereo sound for television, beginning in 1959-60 along with an FM stereo broadcast study, but each inquiry was met with a tidal wave of indifference by the television industry and quietly faded away.

In recent years, the perspectives of both producers and consumers of United States television hardware and programming have been changed by a number of developments. The standardization and nearly universal employment of video tape formats containing multiple audio tracks has been coupled with the improvement of these tracks to reasonable, if not exemplary, quality. Satellite network and syndicated program distribution facilitates the inclusion of multiple high-quality audio channels with the visual signal. The proliferation of high-fidelity stereo consumer audio equipment has spilled over into the video sector, and pre-recorded videocassettes with theatrical-quality stereo soundtracks are as close as the corner video store.

Competition among the video media from such sources as independent broadcasters, cable television, videocassettes and videodiscs, and direct satellite delivery has multiplied. Broadcasters are searching for ways to more fully exploit their television signals, and the color receiver market has matured. These factors have acted in concert to fuel the industry's interest in high-quality multichannel television sound. This brought about re-examination of the MTS question in the early 1980s and led to the development and implementation of the BTSC system for transmitting multichannel television sound, a watershed development in broadcast television audio in the United States.

The rapid acceptance and widespread awareness of broadcast stereo television and the availability of high-quality stereo sound on other video media has spawned efforts on the part of broadcasters and receiver manufacturers to improve the fidelity of television audio from production through the broadcast chain all the way to the living room.

To the public, steréo and high fidelity are so inseparable that they are virtually synonomous. The viewer who is enjoying television in stereo is either using a television receiver with a reasonably high-quality audio section or is routing the stereo television audio through the home high-fidelity system and, in either case, will be only too aware of any fidelity problems in the transmitted audio signal.

The result has been a ripple effect in television audio equipment design and practices which is producing improvements in audio distribution and processing equipment, in aural transmitters, and in television audio systems engineering. New attention is being paid to such factors as distortion, signal-to-noise ratio, and audio bandwidth in both transmitting and receiving equipment.

The audio side of television production is benefitting from the current interest in improving television sound as well. Studio acoustics are receiving new attention. Audio control room acoustics and speaker placement are being scrutinized, and audio consoles featuring impressive performance specifications and highly automated functions are being installed.

A renewed interest in some venerable stereo microphone techniques has surfaced. New versions of the X-Y and M-S type of far-field microphones are appearing which are readily adaptable to television boom miking and field pick-up situations and because they use co-located elements, their stereo signals inherently sum to mono without phase cancellations.

The video tape recorder has been identified as a weak link in the television audio chain, and work is proceeding apace to improve its signal-to-noise ratio, headroom, and distortion characteristics. A prime enemy of stereo television broadcasting is interchannel audio timebase or phase error, which can result from azimuth misalignment of longitudinal audio head stacks.

Changes are being made to current types of video tape recorders to enable routine azimuth adjustment, and some devices have been developed which electronically correct these time delay errors between stereo tape tracks as well. The efforts to correct or eliminate audio timebase errors are critical because such errors degrade the mono sum signal, something no broadcaster can live with.

Along with new video tape formats, some new techniques of recording audio on videotape have recently appeared. These include such developments as noise reduction for longitudinal audio tracks and the incorporation of companded FM audio channels, both of which are available on the new half-inch professional video formats. These FM channels provide wide dynamic range and low distortion, with the disadvantage that they cannot be edited separately from the video.

A development holding great promise for the future of audio on videotape is the employment of PCM digital audio recording for both one-inch and the new MII formats. Sixteen-bit linear quantization and 48kHz sampling provide complete audio transparency, total editing flexibility, multigenerational audio dubbing without degradation, and the absence of time delay or phase errors between stereo channels.

The word in television audio's future is increasingly going to be "digital". The digital trend, already underway, will develop into a major tidal wave, washing over all facets of television audio from production to broadcasting to reception.

In addition to magnetic tape, digital audio will be stored in such media as magnetic disc, solid state memory, and optical disc. In production and post-production, dramatically increasing usage will be made of non-tape based digital audio editing systems, which run the gamut from disc-based emulations of audio cartridge machines to such wonders as the audio sampling synthesizers, which are editing systems and musical instruments integrated into one device. These synthesizers can store and manipulate sound in a dizzying variety of ways, in synchronism with video signals.

Digital audio processing is proliferating in the form of delay and reverberation devices, pitch changers, and time compression devices, all of which are characterized by impeccable audio quality. The future will bring us fully digital audio consoles in which the functions and parameters of audio processing modules will be software-controlled. What the modules do functionally will be limited not by their nature as hardware, but by computer processing power alone; and as an additional bonus, these consoles will diagnose their own faults and reroute signals around malfunctioning sections! As we move into the future, an ever greater amount of the audio editor's work will be done using a computer terminal.

Distribution of audio, both within the plant and to the outside world, will increasingly be done in digital form. The digital domain has some very attractive and useful qualities for television audio, including wide dynamic range, very low distortion, robustness under adverse conditions, and the capability to withstand many generations of recording and many stages of processing and distribution without significant degradation.

The day will come when television audio will remain in the digital domain virtually from its origin to the receiver. Digital audio delivery to the home will give the consumer the ultimate in audio quality from the television set.

The near future will increasingly bring the augmentation of stereo with surround-sound, a natural adjunct to the video presentation, particularly with the expanding presence of large-picture and projection televisions, and most especially to accompany high-definition and wide-screen formats. This will be the next audio step toward bringing the theater experience into the living room. A future generation of television receivers will, no doubt, include integral surround-sound decoders.

In conclusion, the increasing electronic sophistication of the audience, as well as the producers and disseminators of television programming, in combination with the maturation of the television medium and increasingly competitive nature of the video marketplace, have generated and sustained a keen interest in high-fidelity and stereophonic television audio. The cornucopia of new developments in audio for video which are being used and developed by the television industry is creating an exciting future for television audio. At this point, we have only seen the beginning!

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QUICK TV STEREO PROOFS

Bruce E. Hofer

Vice President and Principal Engineer
Audio Precision, Inc.

Introduction

Stereo sound has added an exciting new dimension to the television experience. It has also increased the listener's expectation of quality and the perception of problems. Many people are more sensitive to noise and imperfections in a stereo reproduction than one in mono.

Maintaining a high standard of stereo sound quality requires periodic checks of station performance and adjustments to compensate for equipment drift and aging. Conducting a full proof in compliance with BTSC guidelines is a time consuming process with results that may not always correlate with the true audio quality. A full proof must be conducted during station down time because several tests require the stereo generator to be switched into non-broadcast modes. There is also the natural tendency to put off a proof as long as possible if there are few complaints and it "sounds like stereo."

Recent developments in the field of automated audio test instrumentation have made the concept of an on-air "quick" proof both realistic and practical. A quick proof is an automated sequence of selected tests that can rapidly check the most important aspects of aural quality, under normal station operating conditions. A few carefully chosen tests can detect the presence of almost any problem in the aural chain that would cause a loss in quality. A quick proof could be run weekly, or even daily at sign-on, to provide a periodic record of station performance. Long term drift problems could be spotted before they become audible.

BTSC System Review

A complete description of the BTSC system can be found in the FCC OET Bulletin #60, which is the controlling document for TV stereo in the United States. Many factors led to the specific design of the BTSC system. To provide compatibility with the large installed base of monaural receivers, (L-R) information is added in a double sideband suppressed carrier (DSB-SC) subchannel located at twice the horizontal scan rate, or 31.468 kHz. A 15.734 kHz pilot is inserted to provide the phase reference for demodulating the (L-R) signal in the receiver. All audio information is sharply bandwidth limited at 15.0 kHz to prevent spillover between the main and subchannels and interference with the pilot tone.

The most unique characteristic of the BTSC system is the complex and non-linear processing of the (L-R) sub-channel signal. The simple 75 usec pre-emphasis used in FM radio subchannel processing will not deliver satisfactory results in the TV environment with its nearby strong amplitude modulated video signal. The aural signal is subject to video interference from imperfect bandlimiting, occasional overmodulation, and incidental phase modulation of the video carrier. To overcome these problems the (L-R) subchannel signal is compressed in dynamic range.

Figures 1 and 2 shows a simplified representation of the BTSC encoding process. The (L-R) signal is first pre-emphasized following a different curve than applied to the main channel. It is then compressed in dynamic range and spectral shape. Low frequencies below 1 kHz are compressed by a 2:1 factor while high frequencies above about 8 kHz are compressed by a factor approaching 3:1. Thus, a 60 dB dynamic range in the (L-R) signal will encode to a dynamic range of 30 dB range at low frequencies but only 20 dB at high frequencies. An interesting consequence of this process is that the subchannel modulation will never go to zero in the absence of (L-R) information. Unavoidable noise components located 100 dB below maximum will still cause an encoded subchannel modulation of about -34 dB, or 2%.

The BTSC system also provisions for a second audio program channel (SAP) and a narrow-band channel for station usage (PRO). These add additional carriers to the composite aural signal above the (L-R) subchannel. The SAP channel is encoded with the same processing as the (L-R) subchannel to obtain a reasonable signal-to-noise ratio, but is bandwidth limited to 10 kHz. So far, SAP audio quality has not been the focus of much attention. A typical example of its usage has been for second language commentary of sports events.

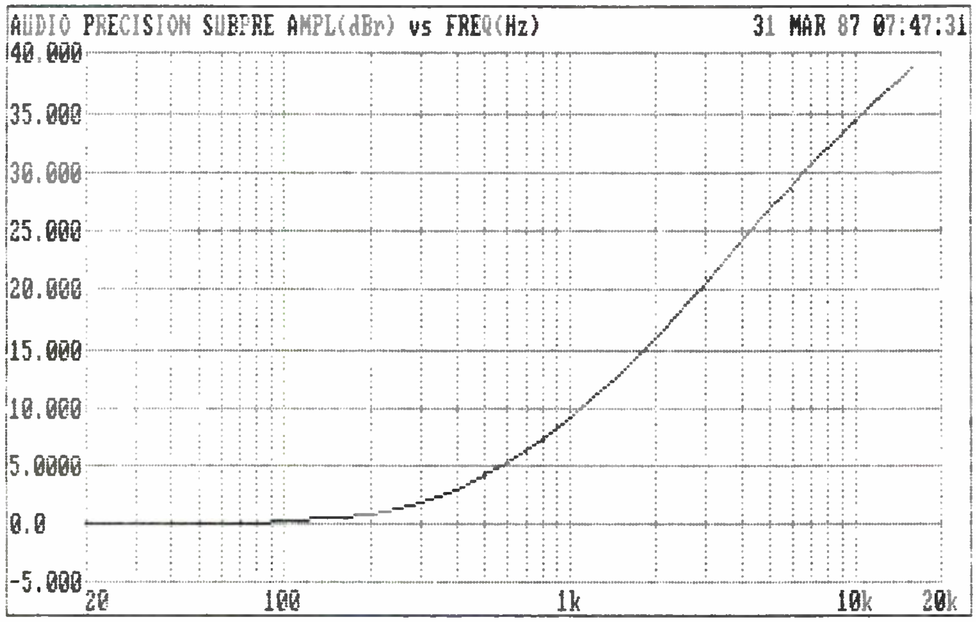


Figure 1 - (L-R) BTSC Pre-emphasis characteristic

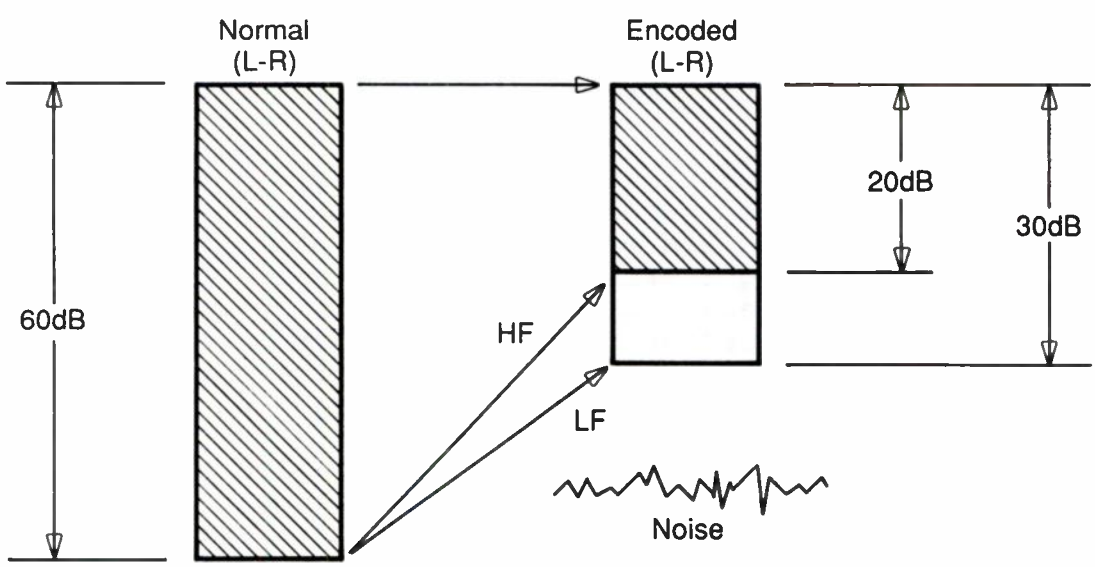


Figure 2 - (L-R) BTSC encoding

Separation Problems

The BTSC system exhibits a very sensitive relationship between stereo separation and processing gain-phase errors. Table 1 shows the degrading effect that a demodulated (L-R) amplitude or phase error will have on separation. To achieve 20 dB separation requires that the total amplitude error in the recovered (L-R) signal, following the encoding-decoding process, be within 1.58 dB. A pure phase error must be less than 11.48 degrees. The combined effect of both types of error on separation is closely approximated by the vectorial summation of each individual effect. Thus, to guarantee 20 dB separation requires the overall (L-R) errors must be less than 1.15 dB and 8.12 degrees, even though each individual error would yield 23 dB separation.

Separation	Max Gain error	Max Phase error
20 dB	1.58 dB	11.48 deg
23	1.15	8.12
26	0.83	5.75
28	0.67	4.56
30	0.53	3.62
32	0.43	2.88
33	0.38	2.57
34	0.34	2.29
36	0.27	1.82
38	0.22	1.44
40	0.17	1.15

TABLE 1 - Separation versus (L-R) gain or phase errors

Extremely high separation is not required for good stereo quality. Many people can detect a slight "narrowing" in a stereo reproduction when separation falls to 20 dB. Indeed, the pleasing stereo sensation is still possible with only 6-10 dB of separation, provided it is constant with frequency. If separation degrades to poor levels at only some frequencies "smearing" of the sound images can occur--the aural equivalent of poor color convergence.

The ultimate reproduced stereo separation will be a combination of both station encoding and home receiver decoding error sources. Because these are uncorrelated the overall effect can vary from the worst-case addition of errors to possible subtraction or even cancellation of errors. The station contribution should be as low as practical. The OET-60 requirement of 30 dB separation is quite reasonable even though only 20 dB might be considered adequate for good stereo.

There are other reasons to partition a total error budget to favor the home receiver. Minimizing the receiver cost and complexity is the obvious one. A more subtle and technical reason comes from the BTSC process.

The encoded (L-R) subchannel signal, itself, controls the BTSC encoding-decoding functions. Filtered versions of the encoded (L-R) signal are used to dynamically control companding and equalization signal processing. Interference or crosstalk into the subchannel following encoding will cause mistracking of the decoding process. An interference product will be misinterpreted by the receiver as a part of the original (L-R) signal and cause a decoding error. This exact error is difficult to analyze and strongly dependent upon the spectral characteristics of the subchannel signal to begin with. (The author has developed a computer program for calculating ideal decoder errors for the special case of a single interference tone, and will make it available upon request.) It should be noted that subchannel interference at -60 dB, near 8-12 kHz, can cause a significant loss of midband separation at 1% equivalent modulation. Quality conscious stations are strongly encouraged to exceed the OET-60 section C(a)12 crosstalk requirement of -60 dB.

Separation can exhibit large changes in value over small changes in frequency. Every station has its good and bad frequencies and it is important to take enough data to get a clear picture of true performance. Figure 3 shows the measured separation at 11 spot frequencies of a stereo generator connected to the baseband input of a modulation monitor through a small attenuator to simulate a subchannel deviation error. Note that the worst midband reading is approximately 38 dB. Figure 4 shows the same setup swept with 81 logarithmically spaced frequencies (80 steps). Separation is clearly seen to degrade to 36 dB near 1.5 kHz.

Finally, it is important to realize that the modulation monitor is not perfect and can contribute its errors when measuring station separation. Typical stereo modulation monitors specify or claim 40-50 dB separation. This can still impart a 1-3 dB uncertainty in a 30 dB measurement. Table 1 shows how a 30 dB separation reading could have resulted from a true station performance of 33 dB and a modulation monitor contribution equivalent to 40 dB separation, assuming worst case addition of pure amplitude or phase errors. However, it is also possible that the true station performance was only 28 dB, if errors were of opposite polarity. Extremely high separation measurements ($\geq$50 dB) should always be treated with some skepticism. They may be the result of a fortuitous cancellation of much larger error sources.

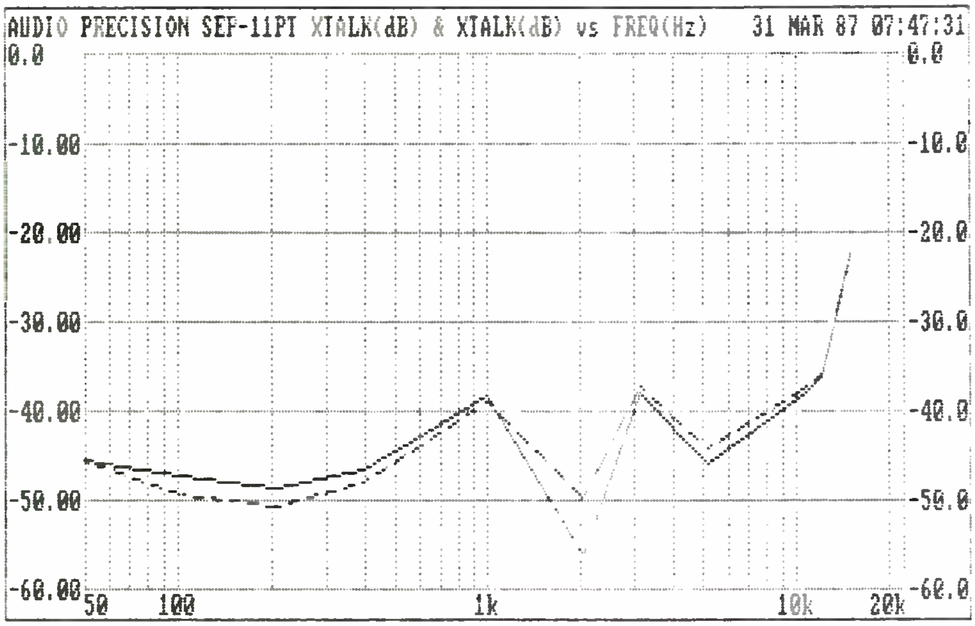


Figure 3 - Separation versus Frequency, 11 points

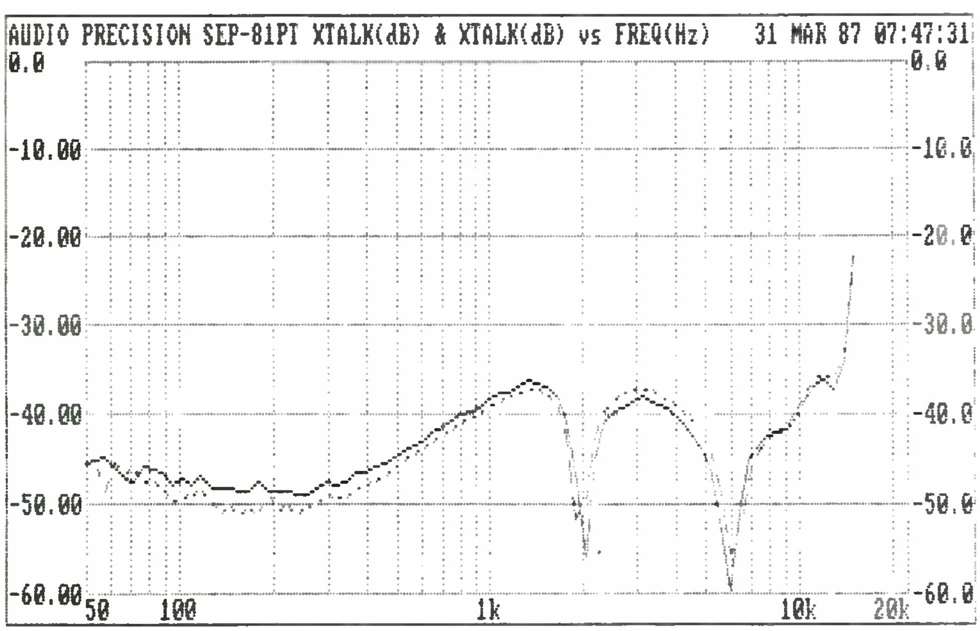


Figure 4 - Separation versus Frequency, 81 points

Other Significant Measurements of Audio Quality

Distortion can arise in both the main and subchannel from different mechanisms. Subchannel distortion can be caused by improper time constants within the companding circuits, particularly at low frequencies. Thus, it is recommended that distortion be measured with both (L+R) and (L-R) test signals. Using an L-only or R-only signal will cause both main and subchannel modulation at a lower level, and the measured THD will reflect a combination of both main and subchannel linearity performance.

The permissible levels of distortion have not kept pace with other high quality audio products. OET-60 allows up to 2.5% THD at midband frequencies. This should be considered an absolutely intolerable amount for high quality. Quality conscious stations should maintain their distortion levels as low as practical.

Besides frequency response, hum and noise measurements also offer one of the best checks that there are no problems anywhere in the audio path. Some audio analyzers have the ability to sweep a 1/3-octave bandpass filter and provide a spectral display of hum and noise components. This is most helpful when diagnosing the source of a problem. Stations should expect that the listener's sensitivity to hum, noise, and distortion problems will increase with the extended response of new stereo receiver loudspeaker systems.

Several Quick Proof Proposals

The acceptance of the quick proof concept depends upon measuring a sufficient set of factors to identify any significant problems in the shortest time possible. A quick proof is not meant to be a substitute for a complete aural proof, but it can provide routine data about station performance that can be used to identify the need for more serious maintenance.

The basic equipment setup requires a programmable stereo audio generator connected to suitable L and R inputs at the studio console or audio switcher. An automated audio analyzer is connected to the L and R outputs of the stereo modulation monitor. One analyzer (the AP System One) is unique in providing a third auxiliary input for using the wideband composite audio output of the modulation monitor. This allows frequency response to be measured at constant total deviation (constant peak composite signal output), and additionally provides the ability to measure (L+R) noise with a precision 75-usec de-emphasis (plus 15.734 kHz notch) option filter.

It is physically attractive to locate the modulation monitor and audio analyzer at the studio and pick the signal out of the air with a high quality demodulator. Because RF signal quality is not always adequate at the studio site the modulation monitor and analyzer may have to be located at the transmitter site and left unattended. There are several ways of automating this "split-site" situation and adapting it to the quick proof concept. The AP System One is an example of an analyzer that interfaces to a personal computer and can be linked via a serial port and RS-232 to a telephone line. The operator located at the studio site would simply dial up the transmitter site and automatically link his local computer with the remote analyzer and assume control. As the proof is running, data would be transferred back to the studio for immediate interpretation, hardcopy, or storage. This is particularly advantageous if something is spotted during one of the tests and the operator desires to perform further checks while he has use of the station. Another type of analyzer uses an FSK signal in the audio path itself to reset or advance the measurement sequence. It does not require a separate phone line, but also can not provide convenient feedback of data.

<u>Quick Proof #1</u> is designed to be a compromise between speed and thoroughness. It takes 69 measurements and will run within 60 seconds using the AP System One. The number and specific frequency points checked can be modified to reduce test time or enable the use of less sophisticated test equipment.

Test 1 - Frequency Response using 50% (L+R) modulation at
30-50-100-300-1k-2k-4k-7.5k-10k-12k-14k-14.5k-15kHz

Test 2 - L and R output THD+N using 50% (L+R) modulation at
50-100-400-1k-5k-7.5k-10k-15kHz

Test 3 - L and R output THD+N using 50% (L-R) modulation at
50-100-400-1k-5k-7.5k-10k-15kHz

Test 4 - Separation using 10% equivalent modulation at
50-100-200-300-500-1k-2k-3k-8k-10k-12k-14kHz

The 50% modulation level of tests 1-3 may be too high (loud) if the quick proof is run during significant viewing hours. Higher modulation levels may trip peak-limiters or other processing devices. The suggested spot frequencies give good coverage of the audio band but may not always be optimum. As mentioned previously, every station has its good and bad frequencies. It may also be desirable to tradeoff the number of distortion check points to increase the number of separation points, particularly near midband.

Quick Proof #2 adds additional tests of stereo separation at 1% and 50% equivalent modulation and a spot test of de-emphasized main channel hum and noise:

Test 5 - Separation using 50% equivalent modulation

Test 6 - Separation using 1% equivalent modulation

Test 7 - (L+R) main channel hum and noise

It requires approximately 50% longer to run than Proof #1. Tests 5 and 6 check the tracking of the encoder and would use the same frequencies as Test 4. Degraded separation at 1% modulation can be an indication of subchannel noise, crosstalk, or interference problems. Test 7 measures hum and noise as mono listeners would receive it and is a good indicator of STL or audio switching problems.

Quick Proof #3 only tests separation (Test 4) and runs in the shortest time. Separation is the single most useful test of station quality due to the complex nature of BTSC subchannel processing. Test time is approximately 20 seconds with the AP System One, but could be further reduced by checking only those frequencies previously determined to be the best indicators of station performance.

Quick proofs based upon other combinations of tests or different selections of test frequencies are possible. Each station will have its own unique set of needs and problems to be satisfied. The complexity, speed, and effectiveness of a quick proof will be limited by the capabilities of the test equipment. The three specific examples outlined above are available today from Audio Precision, but could probably be adapted for use with less sophisticated equipment.

Summary

The quick proof offers the TV station a practical tool to insure consistent high quality stereo. Many aspects of aural performance can be quickly verified by conducting a few selected tests. Recent advances in affordable automated audio test instrumentation have made the concept of a quick aural proof a reality. Viewers will become more critical listeners as both program source material and home receiver quality improve.

THE IMPACT OF PRE-EMPHASIS

By Jerry Carmean

Director of Engineering and Operations
Ohio University, Athens, Ohio

Many broadcast engineers first become acquainted with equalization when trying to measure the frequency response of an audio tape recorder. Not realizing that a record to playback frequency response run must be made at least 17 dB below operating level, a beginning engineer would most likely use normal operating level to make this measurement. At low frequencies there would be no problem but as the audio input frequency is increased, the output level sags and becomes distorted and tones which are not at the input frequency appear. As frequency is increased, the recording equalizer gives an added boost which saturates the tape and these nonlinear distortion products arise.

Such measurement problems not withstanding, equalization is a very important and valuable method of obtaining a greater degree of performance in electronic equipment where there is a frequency-dependent weak link. An excellent example is in the groove placement on a phonograph record. The pickup stylus responds to the rate of change in the position of the groove. This rate of change is greater for higher frequencies (because the length of the wave is shorter) than for the lower frequencies. To get adequate low frequency loudness from the phonograph cartridge, the groove excursion would need to be greatly increased for low frequencies, a practice which would be wasteful of vinyl real estate. A better method (and the method which is in use) is to electrically reduce low frequency levels and increase high frequency levels at a predetermined rate before recording onto disc, then use an exactly opposite frequency characteristic upon playback. This not only reduces required groove excursion at low frequencies, but also provides about 8 dB less high frequency noise in the recording. A slightly different form of equalization is used in the FM broadcast transmission system. This equalization is called pre-emphasis at the transmitter and de-emphasis at the receiver. It is used because of an interesting characteristic of the FM detection system. This will be discussed at a later point in this paper, but first I would like to present some information regarding noise.

Generally the noise power in a communications channel is directly proportional to the bandwidth used in a channel and to the prevailing temperature. The amount of noise is related to our electrical units through a constant called Boltzmann's Constant, K. ($K=1.38 \times 10^{-23}$ Joules per degree Kelvin). A Joule is simply a watt-second and Kelvin is simply the centigrade temperature plus 273^{o}.

How Noise is Generated

An electrical conductor consists of metal or semiconductor atoms in a conductor (wire). The atoms are fixed into place and cannot move around freely. Some of the outer electrons of some of those atoms can break loose rather easily and those electrons are called free electrons because once they have broken loose they are more or less free to move around within this matrix of metal atoms and can easily join with an atom at a different place, or just move on. This is easy to understand if we would consider the wire to be similar to a hollow pipe with boundaries at the surface of the wire. Within this wire, electrons are free to move around, just as molecules of gas move around within a container. Each electron has a negative charge, and this charge cuases an electric field to build up around each electron. These small electric fields repel the fields of each of the other electrons. Each electron in the wire will try to get as far away from every other electron as it can and will intersperse evenly to every part of the conductor. For that reason, you measure the same voltage at every point on a wire if it is not carrying a lot of current. If you inject electrons into one end of the wire (by connecting to a battery or whatever) each of those electrons squeeze into the crowd and put force on the other electrons with its own electric charge. More electrons in the wire represent a greater electron pressure which will resist the injection of more electrons, and a stable state is quickly re-established. A current takes place when electrons are injected into one end of the wire and allowed to leave the other end. Again, each electron is trying to stay as far away as possible from every other electron because of their electric fields. Notice that the electric pressure is due to the electric fields of the electrons which are forced to stay within the boundaries of the wire. A difference in electric pressure is voltage, and when two wires are connected, this pressure will try to equalize because each electron tries to get as far away from each other electron and still stay within the boundaries of the wires. This rearranging of electrons in response to a voltage (pressure) difference is called current.

In addition to the forced movement of electrons in a wire there are random motions due to temperature. Heat is simply motion of atoms or electrons, and every temperature has a corresponding degree of motional activity. These electrons are bouncing off the walls of the copper container and rebounding from each other's electric fields within the container. A movement of an electric charge is a current and this random motion of electrons within a container (wire) represents random small currents. The magnitude of these currents is proportional to temperature because higher temperatures mean faster moving electrons.

Since these electron motions are random, currents at all frequencies, or white noise is implied. The wider range of frequencies that are measured, the more noise power is included in the measurement.

If this noise is developed across a resistor, the voltage across this resistor is

$$V = \sqrt{4KTR} \qquad \text{VOLTS}/\sqrt{H_Z}$$

and if the voltage is considered over a specific bandwidth, then

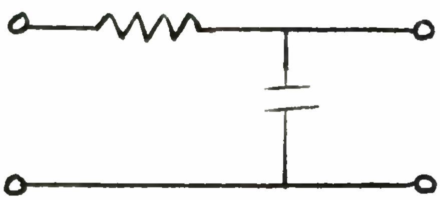
$$V = \sqrt{4KTRB} \qquad \text{VOLTS}$$

V = RMS VOLTAGE ACROSS THE RESISTOR
K = 1.38×10^{-23} J/K
R = RESISTANCE IN OHMS
T = TEMPERATURE IN DEGREES KELVIN
B = BANDWIDTH IN H_Z

Noise in a communications system behaves in the same ways as noise generated by a resistor. Noise power is directly proportional to bandwidth. Noise voltage in an AM detector is proportional to the square root of the bandwidth. In a wideband FM detector, however, the noise voltage is directly proportional to the bandwidth, not the square root of the bandwidth. It is, therefore, quite advantageous to use a lowpass audio filter after the FM detector to reduce this added noise.

Maintaining a flat frequency response with this low pass filtering requires a complementary high pass filtering in the transmitter prior to modulation of the transmitter. The net result of the use of the pre-emphasis and de-emphasis is an unchanged frequency response. If the de-emphasis network is a dual of the pre-emphasis network, there will be no change in phase with respect to frequency.

The pre-emphasis and de-emphasis curves are usually generated using a passive resistor-capacitor or resistor-inductor network connected as a single pole or single zero low or high pass filter. In its simplest form, a de-emphasis network looks like this:

IN  OUT

Some possible Values of C and R

C	R	RC
75 uf	1 ohm	75 us
0.75 uf	100	75 us
.0075 uf	10,000	75 us
75 pf	1 meg	75 us

Figure 1. Simple Pre-emhasis Circuit

Any combination of resistance and capacitance values can be used as long as the capacitance in Farads multiplied by the resistance in ohms equals 75×10^{-6}. The corresponding pre-emphasis circuit is somewhat more complicated, requiring a load impedance which is very low with respect to R.

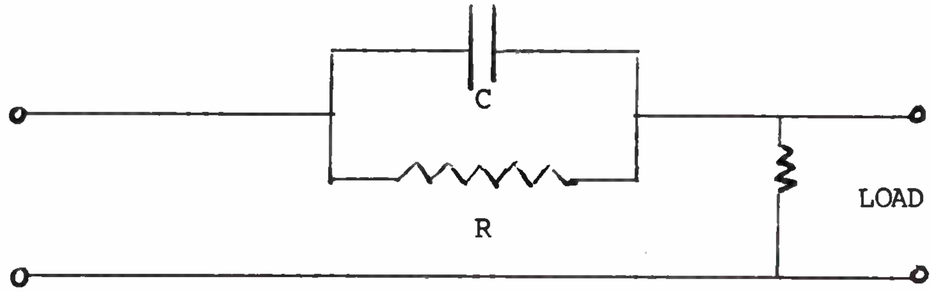


Figure 2. Simple De-emphasis Circuit

Operational amplifiers make simple pre-emphasis and de-emphasis circuits easy to achieve. The graph of the frequency response of these networks is shown below:

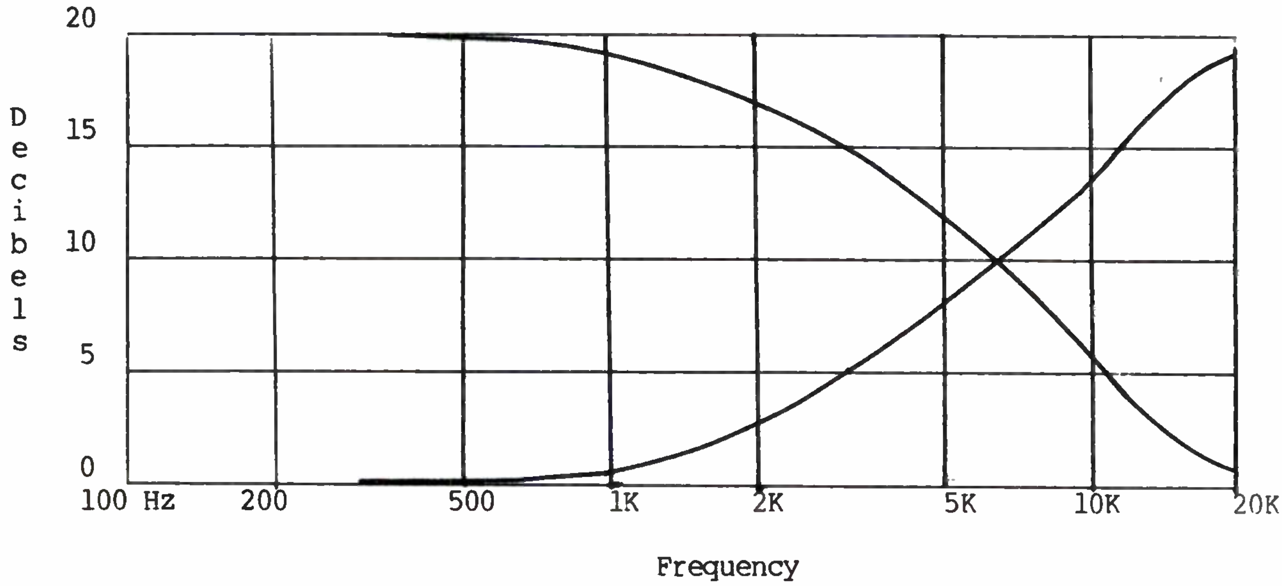


Figure 3. Pre-emphasis and De-emphasis Curves

Note that the effect of pre-emphasis and de-emphasis is minimal at low frequencies but increases to 13.6 dB at 10 KHz and to 17 dB at 15 KHz. This can cause problems when you are dealing with limited headroom or when it is necessary to do some signal processing. For example, how can you do simple limiting to prevent overmodulation of a transmitter when pre-emphasis is used? If you run the signal through a limiting amplifier before pre-emphasis, the high frequencies can still overmodulate the transmitter because of the high frequency boost after the limiting. If the overall modulating level is reduced to prevent high frequencies from overmodulating the transmitter, then your modulation meter will read about 30% modulation at the same time your 100% peak flasher is flashing. One early solution to this problem was to limit the signal <u>after pre-emphasis</u> or to pre-emphasize the signal, limit the signal then de-emphasize the limited signal again before putting it into the transmitter, where it was pre-emphasized again. this is how the CBS Volumax of 20 years ago managed to get a high percentage of modulation with a pre-emphasized signal. This process, however, gave a dynamic dulling of the high frequencies which was not particularly pleasant to listen to. Modern processor-limiters for FM have done a remarkable job of giving a wide-band, clean sound with a relatively high percentage of modulation.

Measurements of audio characteristics in a pre-emphasized transmitter are also a little more complicated and quite tricky to the uninitiated. For example, as you measure the frequency response of an FM transmitter at a constant level of modulation you find yourself needing to lower the audio input to the transmitter as the frequency increases. By the time you have reached 15 KHz, the input to the transmitter is 17 dB lower than it was at 400 Hz and the output of the modulation monitor has also dropped by 17 dB. This is not a result of sloppy measurements, but rather, a consequence of using pre-emphasis in the transmitter and de-emphasis in the modulation monitor.

Another interesting situation you may run into is an apparent increase in the harmonic distortion as the frequency is increased. This distortion increases as the percentage of modulation is decreased. This distortion isn't real, it is simply the result of the greater system noise reaching the distortion analyzer. The reason for this noise increase is that the signal input to the console is reduced (remember the 17 dB at 15 KHz) while the gain of the console and all other amplifiers remain the same. Harmonic distortion is measured by calibrating the tone input level, cancelling out the tone and calling everything which remains, "distortion". The distortion analyzer shows this noise on the meter. The FCC recognized this problem when setting up rules for doing FM Proof of Performance Measurements and did not require distortion measurements at 50% and 25% modulation at frequencies above 5000 Hz.

Pre-emphasis With AM

Now, let's take a look at reasons for using pre-emphasis on AM. Remember, that the noise voltage output in an AM detector is proportional to the square root of the system bandwidth, so the use of de-emphasis and corresponding pre-emphasis will not improve the signal to noise ratio. Gaining a better signal to noise ratio is not a reason to use pre-emphasis on AM.

Think for a moment. At what modulating frequencies does your antenna present the worst load to your transmitter? At what modulating frequencies is your transmitter least able to deal with a difficult load? Are these the frequencies that you really want to boost?

The National Radio Standards Committee, sponsored by the Electronic Industries Association and the National Association of Broadcasters has been working on the problem of second adjacent channel interference. Harrison Klein and Hammett & Edison have recently undergone a study of overmodulation and excessive bandwidth under the sponsorship of the National Association of Broadcasters. Both groups have published very informative and important reports dealing with ways to improve AM broadcast. I strongly urge you to get a copy of both of these reports and study them carefully. Obvious in both reports is the necessity to reduce the spectrum bandwidth. NRSC suggests the use of rather severe low-pass filtering to remove all audio frequencies above 10 KHz, and this appears to be in line with Harrison Klein and Hammett & Edison's report. This filtering should substantially reduce or completely eliminate second-adjacent channel interference if this were used by all stations. The NRSC suggests brightening the sound by using a combination of the 75 uSec pre-emphasis as is used in an FM transmitter in addition to an 18.3 uSec de-emphasis circuit followed by a brickwall, low pass filter to remove all audio above 10 KHz. The pre-emphasis, de-emphasis and brickwall filter are all in the transmitter processing system. The reason for using the pre-emphasis circuit is that "Everybody's doing it." I do not agree that this is a good reason for using pre-emphasis. The 18.3 uSec de-emphasis circuit is used because the pre-emphasis circuit is too severe. It will reduce the 13.6 dB rise at 10 KHz to a 10 dB rise. A brickwall low pass filter is necessary partly because of the boost at 10 KHz. Otherwise, a less severe filter could be used.

Severe filtering gives considerable amount of phase shift at frequencies near the cutoff frequency and if the filter is not very carefully designed excessive phase shift could allow overshoots to overmodulate the transmitter. This overshoot problem would be aggravated by pre-emphasis.

In closing, I urge you to look at both the NRSC report and the Harrison Klein report. Familiarize yourselves with NRSC's proposal and help clean up the AM band. I would also urge the NRSC to seriously consider reducing the amount of pre-emphasis immediately, or at the yearly review periods. The revelation that many AM stations are trying to use pre-emphasis in excess of 75 uSec and that NRSC is suggesting the 75 uSec curve as a method of <u>reducing</u> the amount of pre-emphasis commonly being used was apalling to me. If your station is using excessive pre-emphasis, spend time thinking about how much it is hurting other stations which are not in your market and just how much it is deteriorating your own signal because of the distortion which is generated in your processor and in your transmitter.

THE ROLE OF A ROUTING SWITCHER IN A RADIO STATION

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Radio is changing. In the past 5 years it has become more competititive, more dynamic, and more responsive. Stations are being sold and moved, with formats and calls changing almost daily. A multitude of new programming sources are finding their way into our facilities, from satellite channels to remotes and call in shows. These changes are resulting in changes in the way we as engineers work as well.

In the past, when a studio was remodeled, or built from the ground up, the engineer developed the configuration of the new studios by analyzing the station or AM/FM combination and its format and projecting future needs. The new station may not change appreciably for 5 to 10 years. This process often fails today as the format may be changed before the construction project is complete. What is needed is a method which allows fast and easy reconfiguration of a station.

When the television industry faced a similar problem of mushrooming sources and changing requirements some years ago, they turned to a technology pioneered by the phone company, routing switchers. These electronic marvels can switch our telephone calls quickly and accurately, from millions of phones in use worldwide. This development revolutionized the telephone business, and the operator with her rows of plugs and jackfield are gone forever. In television, the switcher also changed the business, and allowed tremendous flexibility and complexity with a minimum of confusion and error. Hundreds of input sources like cameras and consoles are switched to hundreds of output destinations like monitors and VTR's.

What is a routing switcher ?

Basically it is a group of switches arranged such that any input may be connected to any output or group of outputs. The manner in which the switches are arranged is often known as a matrix. A matrix can be visualized as a rectangle, with the inputs connected to the vertical side and the outputs along the bottom. Where the lines cross is known as a crosspoint, the actual switch. This crosspoint may either be open or closed, and thus any horizontal line (input) may connect to any vertical line (output). The number of switches required by any switcher is equal to the number of inputs multiplied by the number of outputs. As shown in Figure 1, the rest of a routing switcher merely augments this basic matrix.

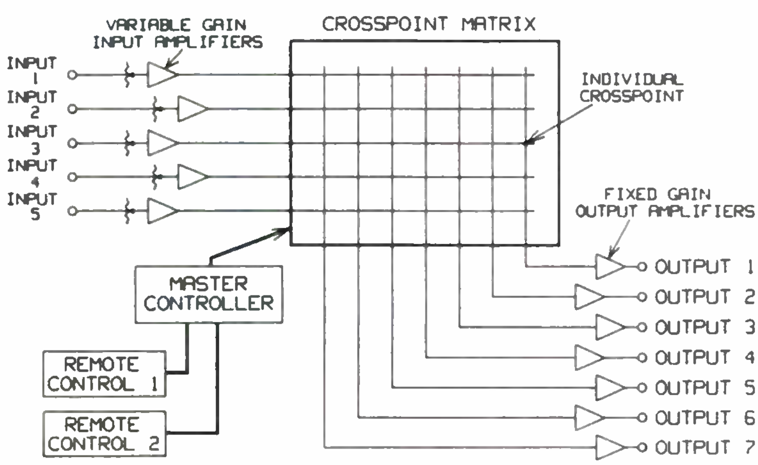

FIGURE 1.

In stereo systems, the mono matrix is doubled, so that a left and right matrix exist, and they are switched in parallel by the controller. This configuration, shown in Figure 2 is ideal for all stereo systems, but radio stations seldom are totally stereo. Even in FM stand alone operations a considerable number of mono inputs often exist such as phone lines and remote news RPU's. When it is necessary to connect a mono input to a stereo dual matrix, the input must be connected in parallel to both the left and the right matrices. This problem gets worse in AM/FM combination facilities.

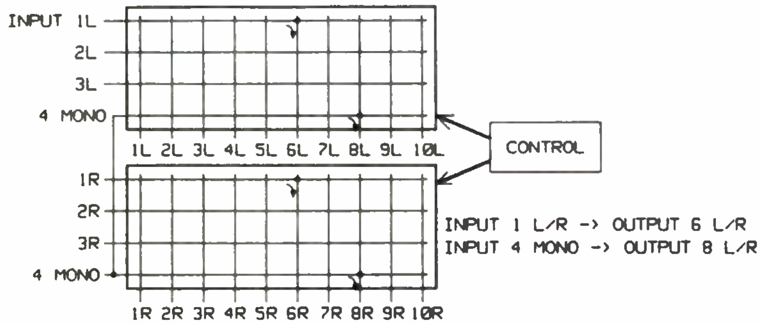

FIGURE 2. STANDARD STEREO DUAL MATRIX

A more efficient and flexible system is shown in Figure 3. Known as "Wild Audio" it is a combined configuration of the left and right matrices into a single matrix. With this matrix, mono inputs need only be connected once, and the controller is programmed to route the mono input to both left and right outputs when connection to a stereo output is desired.

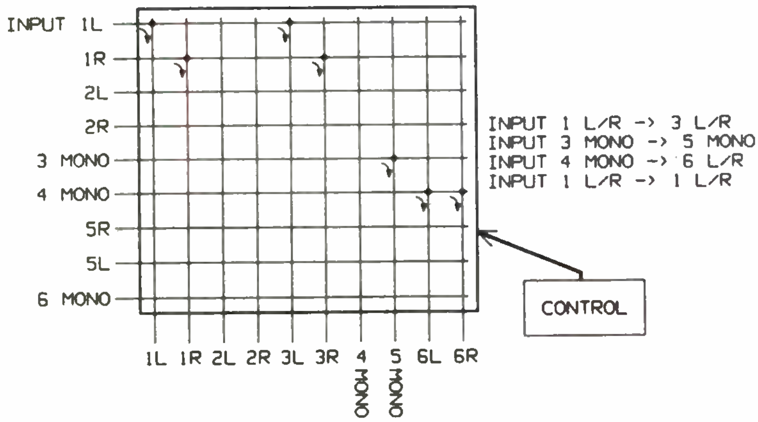

FIGURE 3. WILD AUDIO STEREO/MONO MATRIX

The technology of the actual crosspoint switch is very diverse. Many smaller switchers employ relay switches, and some of those are magnetically latching. The advantage of this approach is that if a power failure occurs, and the system is passive (without input or output amplifiers), the audio will still be routed to the proper outputs without interruption. Other active switchers employ various semiconductor switches in the crosspoint. A key feature to look for with active crosspoint switchers is non-volatile memory, assuring that, on power-up, the crosspoints will be in precisely the same configuration as they were when the power went off. In the past few years, Field Effect Transistor switches have become popular due to their high reliability and good audio performance. Some manufacturers have integrated the crosspoint circuitry into large scale integration or hybrid circuits to reduce size.

In many switchers, input amplifiers before the matrix isolate the inputs and provide gain adjustment. Maximum dynamic range is assured in a system where all inputs are adjusted to a standard level. Most modern switchers provide a balanced, RF protected, variable gain input. If the variable gain input can accommodate a wide enough range, most of the distribution amplifiers in the facility may be eliminated.

Following the matrix an output amplifier is required to prevent the loading of any output from changing the level of any other output being fed from the same input. Generally these output amplifiers also provide balanced outputs capable of driving the levels and loads common in radio today.

The most complex and potentially the most beneficial component in the switcher is the Master Controller. This module, as its name implies, is responsible for controlling all of the matrix switches, and thus the signal flow in the station. The benefits of the Master Controller come from the intelligence and remote control the unit provides.

The Remote Controls are the operator interface to the switcher. They allow the operator to select the inputs which will be routed to the outputs in his studio. There are many types of remote controls available, ranging from simple thumbwheel switch assemblies to intelligent alpha-numeric displays with plain English readout of the selected inputs. The selection of the remote controls is an important one, as operators generally have a difficult time understanding traditional patch bays, and the remote control becomes an ideal opportunity to select an alternative which significantly reduces the confusion and errors on air.

There are many methods of accomplishing the remote control connection to the master control. Some units require a coaxial cable or a multipair cable carries a BCD logic signal, while others utilize existing twisted pair audio cable.

Most switchers offer redundant power supplies which automatically back each other up in the case of a failure. A good idea is to have these supplies fed from different phases of station power, assuring the switcher will remain operational if a phase is lost.

How do I use a Switcher in my station ?

A routing switcher is most effective when it is connected to all inputs which are used in more than one studio or by than more than one output. Typical inputs and outputs from a switcher are shown in Table 1-1. In this way, the switcher replaces all distribution amplifiers and recorder input selectors, most patch bays, and reduces considerably the need for large and complex consoles as well as minimizing the amount of station wiring needed. Not all patch bays are normally replaced, as the most critical circuits usually are backed up via patch bays in order to reduce the chance of a system failure.

TYPICAL SWITCHER INPUTS
NETWORKS
LOOPS
REMOTE PICKUP UNITS
TELEPHONE INTERFACES
CONSOLE OUTPUTS (BOTH PROGRAM AND AUDITION)
AIR MONITORS

TYPICAL SWITCHER OUTPUTS
CONSOLE INPUTS
REEL TO REEL AND CARTRIDGE RECORDERS
TELEPHONE FEEDS
PROGRAM DIRECTOR/ NEW DIRECTOR OFFICES
TECHNICAL SHOP
TRANSMITTERS (PRIMARY AND BACKUP)

TABLE 1-1.

As opposed to conventional patchbay and distribution amplifier wiring methods, the switcher is easily expandable. If a new satellite dish appears in the back yard, the switcher grows with the addition of an input amplifier and matrix elements, instantly providing the new source to all outputs in the station.

Generally, the switcher is located in the main wire room, where all the cables are run from the various studios and where the telephone demarcation blocks are. This minimizes the wiring needed to add new inputs to the system. Methods of connection to switchers range from simple connector blocks to various connectors and even umbilical cables to punch blocks or "Christmas trees". It is advantageous to employ a connection scheme which facilitates quick additions of inputs and outputs while providing identification of the wires and terminals.

What can an intelligent Switcher do for me ?

Many of the new generation of switchers are intelligent, in that they are capable of doing more than simply making a crosspoint change in response to a BCD remote control. These features allow a tremendous advance in the control, flexibility and efficiency in today's radio stations.

One of the most powerful functions is the Salvo. A salvo is a list of instructions to the matrix which can be executed automatically, and nearly simultaneously. These instructions allow a reconfiguration of a studio, for example, from a music show to a sports program. These types of changes can be quite extensive, and a single errant patch cord can often ruin a program. With the salvo function, the reconfiguration can be memorized by the switcher, and may be invoked at any time.

Many Switchers include a clock / calendar which, when used with pre-programmed salvos may cause changes to occur at certain times and/or days. For instance, if a particular program always occurs at a particular time on Saturday afternoon, the switcher can memorize the configuration and time and automatically reconfigure the studio at the proper time, as well as memorizing the configuration for the program which follows to allow a return to normal programming when the ball game is finished. Usually, many salvos may be stored, the total number depending on the amount of storage each requires. This preprogramming may be done at any time, reducing the need for the engineering department to be on hand during a particular program.

Salvos may also, in some systems allow automatic control of a function called "Machine Control". This function consists of a number of relay or opto-isolator contacts under control of the switcher. These may be used to start a recorder to record a particular program at a certain time, or cause any other action which is remotely controllable by a contact closure.

Conclusion

The Audio Routing Switcher represents a technology which can be used to significant advantage when new design and construction is planned. It often is more cost effective than the traditional patch-bay and distribution amplifier method, and allows considerable flexibility in both day to day operations as well as in the changes and growth normal in today's radio.

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AUTOMATIC AUDIO TIMEBASE CORRECTION

By Kristi Urquidi

Senior Engineer, Howe Technologies Corporation

Abstract

Stereophonic television/video production, duplication, and transmission are prone to the occurence of unintentional time delay errors between stereo channels. These errors shift the apparent spatial location of the stereo image and may seriously degrade the monaural frequency response. The cross-correlation technique of phase error measurement and correction is discussed. Sources of time delay error are identified and an error correction device which operates automatically in real time and its applications are described.

Introduction

With the advent of CD's, laserdiscs, and VHS HiFi, broadcasters of stereo television and radio programming are compelled to provide audio material of comparable quality. A serious but often overlooked problem is that of interchannel time delay errors. When stereo signals are mixed into mono, severe frequency degradation can result if spurious time delay errors are not first corrected. Because most viewers listen to stereo material on monaural receivers, the market impact of substandard transmission and reception can be significant. The problem of mono capatibility therefore becomes a serious economic issue.

As stereo television gains momentum in the industry, the problem of differential time delay error must be addressed. It has been shown that cross-correlation is an effective technique of measuring interchannel phase error in stereo audio signals. Proprietary circuitry based on cross-correlation allows real-time correction of time delay errors, providing mono compatibility without compromising the accuracy of the original stereo image.

Time Delay

When, in a stereo audio signal, an unintentional differental time delay is introduced and the signals remain discrete, the apparent stereo image shifts away from the delayed side. Time delays less than 50 microseconds produce image errors which are almost negligible. Delays greater than 100 microseconds begin to yield perceptible image shifting. When the two signals are summed into mono, however, relatively short time delays (from 10 to 100 microseconds) lead to audible artifacts. This is due to the comb filter effect, which produces a number of null points in the frequency response spaced at regular harmonic intervals, where the first null occurs at a frequency having a period equal to twice the delay.

For example, assume that channels "A" and "B"carry signals composed of five separate tones at 1000 Hz, 2000 Hz, 3000 Hz, 4000 Hz, and 5000 Hz. Let 1000 Hz be the fundamental frequency, with the next four tones being the second, third, fourth, and fifth harmonics. (Refer to Figure 1.)

Channel B has been delayed in time by delta t, 500 microseconds (half the time required for the 1000 Hz tone to complete one cycle). As shown in Figure 1, the 1000, 3000, and 5000 Hz signals on channel B have been shifted such that they are inverted images of the corresponding signal on channel A. This phase shift, which is 180°, should not be confused with a polarity reversal, however. If channels A and B were summed to form a monaural mix, the net result would be a complete cancellation of the fundamental and the higher order odd harmonics. The higher order even harmonics remain phase coherent (although boosted in amplitude by a factor of two); this yields the comb filter effect. Other non-harmonic signals would be attenuated by intermediate amounts, depending on the frequencies. As the differential time delay increases, the periodic amplitude nulls will be located at progressively lower frequencies in the audio spectrum.

Detection and Correction of Time Delay Errors

Given a stereo audio signal, the relationship between the left and right channels can range from non-correlation to complete correlation (equivalent to mono). It has been found that the correlation coefficient falls between these two extremes, and in fact has a value approaching 1.0. It is this significant proportion of correlated material which leads to the listener's perception of a stereo image.

It has also been found that well-produced audio has a stereo image which is, on the average, balanced around a spatial center point. These two factors--a high degree of correlation and a balanced stereo image--provide the basis for the error detection and correction scheme.

By using a negative feedback loop topology sensitive to interchannel time delay errors, a correction factor can be derived and applied to restore the original time relationship between the channels. Extensive studies have shown that a full spectrum cross-correlation can often lead to unsatisfactory results. Therefore, the input signals to the cross-correlator are precisely band-limited and amplitude controlled. In addition, a proprietary 'window of zero correction' circuit accurately discriminates between normal stereo phase fluctuations and constant systematic time delays: the stereo information is left intact.

Circuit Description

The negative feedback loop signal is produced by a standard analog cross-correlator circuit, followed by the proprietary 'window of zero-correction'. The output of the cross-correlator, representing the instantaneous phase difference between the input channels, drives a pair of integrators (with different time constants) operating in parallel (see Figure 2).

When the output of the fast integrator exceeds the threshold of the window of zero correction, the time delay control signal changes the amount of applied correction very rapidly, such that almost all of the error is eliminated. Just prior to achieving complete correction, the fast integrator is automatically disabled, thus preventing any overshoot or ringing. The slow integrator then fine tunes the control signal, gradually removing the small amount of remaining time delay error.

The time constants of the two integrators have been selected so that the phase fluctuations inherent in a normal stereo signal (which give the signal its apparent "width") are not affected. This design allows rapid correction of large-scale time delay errors without compromising the original signal's integrity. In addition, the time delay networks have been designed so that the correction applied is a linear function of the control signal, providing a convenient means of monitoring the error in real time.

Applications

The various pieces of program material which make up the final audio tracks of a film or videotape--background music, dialog, sound effects, etc.--must pass through several complex systems before the final product reaches the consumer. Time delay errors can potentially occur during any stage of this process.

And, as these errors are cumulative, the resultant program material may become both phase incoherent and mono incompatible. In order to maintain the original phase relationships throughout this process, it is essential that audio timebase correction be effectively applied in all aspects of the audio-for-video chain: production, post-production, duplication, and transmission.

Production and Post-Production

During the original production and recording of a program's audio "components" listed above, time delay errors can be introduced by such factors as poor microphone placement technique, the utilization of low quality pre-amplifiers and mixers, and improperly configured signal routing equipment.

Throughout the subsequent production and post-production activities, each of these audio components may be re-recorded numerous times. Whenever audio signals are transferred from one recording medium to another, however, various types of interchannel time delay errors are inevitable.

Magnetic tape systems are particularly susceptible to these types of errors. Three examples of their possible cause are: (1) the wide variations which exist in the tape head azimuth and pole piece gaps from one analog tape machine to another; (2) head alignment techniques which differ from one engineer to another; and (3) the lack of correlation between the phase relationships of the "leader" alignment signals on a tape and the program material which is subsequently recorded on the same tape.

Furthermore, when multi-channel program material is mixed down from multiple sources, it is possible to mix the time delay errors as well: this results in an uncorrectable problem. It is only by correcting such time delay errors prior to the mixdown that such unresolvable problems can be prevented. The simultaneous mixing of sound effects from various mag tape machines, for example, can result in the "layering" of unrelated time delay errors on the master tape.

In addition, because the components of a final program are often produced and assembled at different locations and on different equipment, it is possible to have the overall time delay error change at the various splice points. A cross-correlation correction device that detects and corrects these errors in real time is necessary to insure that the final program material is consistently phase coherent and mono compatible.

Duplication

Once a final audio mixdown has been completed and transferred to the master, duplication for distribution is performed. The same causes of time delay errors which plague the production and post-production stages also affect the duplication process.

In order to produce high quality, phase coherent transfers, it is necessary to align the source machine properly for each new master tape. Since tape head alignment is subject to both misadjustment and mechanical drift, even regular calibration is often not enough. The installation of a cross-correlation correction device between the source and the duplication machines is an insurance policy to minimize these errors, improving the overall quality of the copies.

Transmission

Unlike duplication facilities, broadcasters are particularly vulnerable to errors not of their making; they do not have the time to calibrate the station's playback equipment individually for every program aired (especially if it's a live program feed). Even if strict engineering practices are upheld and adjustments are performed regularly, time delay errors may have already been introduced during the production, post-production, and duplication of the programs. Unfortunately, broadcasters have very little control over the integrity of any program material not created in their own facility.

Discrete stereo transmission requires two parallel audio paths, having identical electrical characteristics, so any program materials received via satellite or STL's should also be monitored for time delay errors. If the processing electronics for either path introduces any time delay errors, phase coherence and mono compatibility of the signal which reaches the viewer will be degraded.

In either case, the proper installation of a real-time error correction device in the audio chain will minimize the negative effects of the ever-changing time delay errors inherent in the various programs transmitted!

It should be noted that frequency dependent differential phase shifts appear to the correction device as continuously varying time delays. An example is the output of a comb-filter based stereo synthesizer. It is therefore recommended that the correction unit be installed in the audio chain prior to any such device.

The Consumer

With the audio consumer's increased level of sophistication, the importance of improving the quality of the transmitted signal to the millions of monaural television sets around the country is obvious.

Further, with the proliferation of MTS stereo television transmission and reception equipment, consumers are becoming more discriminating in their appreciation of quality audio program material. Accurate reproduction of the audio signal at the consumer end is crucial....the early detection and correction of time delay errors will insure proper stereo imaging, thereby maintaining the artistic intent of the program's original producer.

The television consumer does not live by bread alone, however: the influence of the motion picture industry on his awareness of quality audio cannot be discounted. In particular, the Surround Sound system (developed by Dolby Laboratories), has had a significant impact on the consumer's appreciation of multichannel audio and its enhancement of the audio/visual "experience".

As Surround Sound audio enters the home environment on videotapes, laserdiscs, and broadcast television, the consumer will come to expect theater-quality audio with his own equipment. Unintentional time delay errors in the original program material or the transmission process, however, can often result in improper decoding of the Surround Sound signal. This often results in center channel information leaking to the surround speakers in a severely distorted form. Time delay errors in the phase encoded matrix will be properly corrected by the time delay correction device, insuring that Surround Sound program material (1) is mono-compatible; (2) is reproducible in stereo with no compromise of the audio quality; and (3) that, when decoded, it accurately reproduces the intended imaging effects of the original multichannel program material.

Summary

The audio timebase corrector based on cross-correlation provides the ability to eliminate undesirable time delay errors in the production, post-production, duplication, and transmission of stereo programs. It has the additional advantage in its ability to be compatible with, and to correct time delay errors in, Dolby Surround Sound encoded program material. This device is thus an indispensable diagnostic and correction tool throughout the television and radio industries in their move toward higher quality audio programming.

References

[1] Ancona, E., "**Practical Problems in the Production, Handling, and Transmissions of Stereo Audio for Television**", Audio Engineering Society 4th International Conference, pp. 180-186 (1987 May).

[2] Hart, A., "**Stereo Audio Transfer Considerations: Some Observations and Thoughts**", Audio Engineering Society 4th International Conference, pp. 117-121 (1987 May).

[3] Hoffner, R., "**Stereo/TV Transmission: Monitoring and Mono Compatibility**", Audio Engineering Society 4th International Conference, pp. 95-101 (1987 May).

[4] Howe, D., "**Stereo phase Error Detection and Automatic Phase Correction Using an Audio Cross-Correlation Technique**", 1985 NAB Engineering Conference Proceedings.

[5] Julstrom, S., "**A High-Performance Surround Sound Process for Home Video**", Journal of the Audio Engineering Society, vol. 35, No. 78, pp. 536-549 (1987 Jul/Aug).

[6] Laletin, W., "**Time Delay and Phase Error Detection and Correction**", 1987 NAB Engineering Conference Proceedings, pp. 119-126.

[7] Murphy, S., "**Live Stereo Audio Production Techniques for Broadcast Television**", Audio Engineering Society 4th International Conference, pp. 175-179 (1987 May).

[8] Schulein, R., "**A Compatible Surround Sound System for the NTSC-MTS Stereo Television System**", 1987 NAB Engineering Conference Proceedings, pp. 207-212.

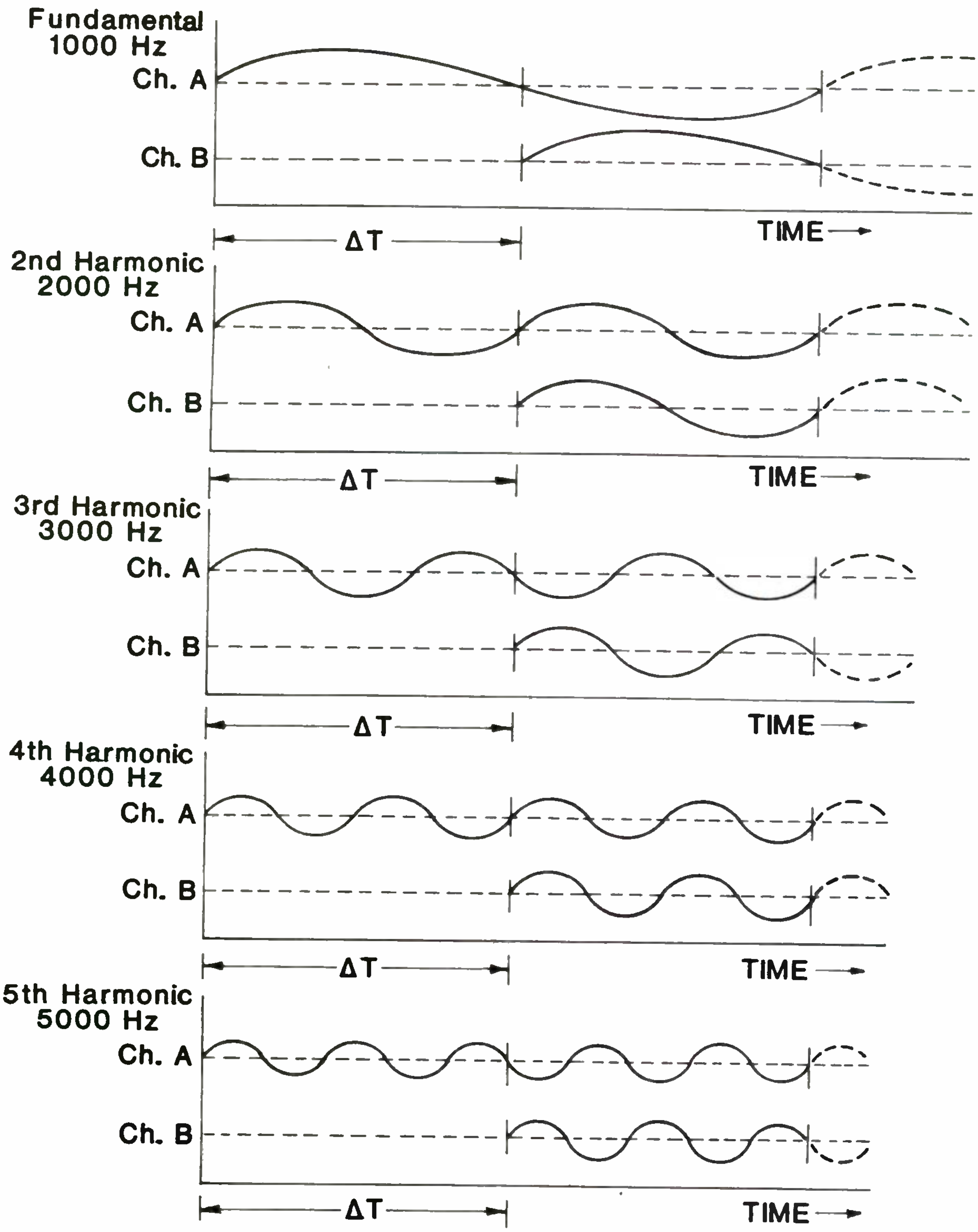

Figure 1. Real Time Phase Error Correction.

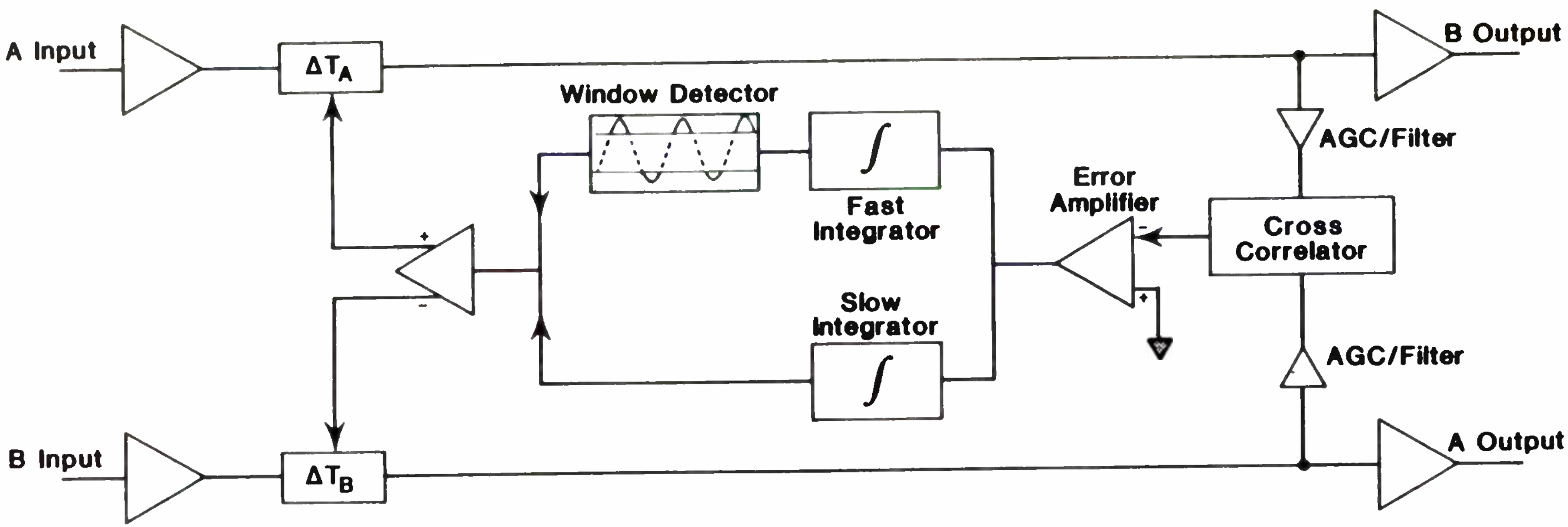

Fig. 2. Audio Time Base Corrector.

TAPE CARTRIDGE TRANSPORT SYSTEMS:

DESIGN, OPERATION AND MAINTENANCE

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INTRODUCTION

The tape cartridge transport system is the key mechanical element of today's modern cart machine. The purpose of the transport system for NAB (National Association of Broadcasters) type tape cartridges is quite simple: support the cartridge and pull the magnetic tape across the heads at the proper speed. Accomplishing this task requires knowledge and technical expertise in many areas of design. Proper design engineering and testing of the many components that comprise this system are paramount in producing a functional, reliable product.

This paper will discuss the design philosophy and specifics of each of these components. Items discussed include the deck plate, solenoid, drive cable, drive cam, cross shaft, pressure roller shaft, pressure roller, motor capstan, cartridge guides and head box.

The functional operating relationships between these components that make up the transport system will be reviewed. A free body force diagram will detail the relationships between solenoid force, pressure roller force and tape pull force.

And finally, a section covering the proper maintenance of this system will be included.

NOTE: All of the design, operation and maintenance data discussed in this paper are in reference to Broadcast Electronic's new "C" series tape cartridge transport system.

COMPONENT DESIGN

Deck Plate - The deck plate (see Fig. 1) is the foundation of the tape transport system. It is the platform that contains all of the other assembled components. The deck must be strong, stable, yet simple in design. It must functionally position the right hand cartridge guide, the head box, the pressure roller and motor capstan in accordance with NAB specifications. Proper placement of these items will assure workable interaction with NAB designed tape cartridges. To maintain the critical locations of components, the deck must be correctly dimensioned and tightly toleranced. Tolerance levels of +/- .002" are quite common. The best material for deck design is .500" thick aluminum tool plate (6061-T651 Type 200) which has excellent strength, stability and surface flatness characteristics. To maintain proper perpendicularity between the deck surface and the motor capstan, a high degree of flatness and parallelism must exist between both the top and bottom surface of the deck. Readings of only .005" TIR (Total Indicator Reading) for both flatness and parallelism must be maintained. The top surface roughness of the deck should not exceed 32 microinches (1 microinch = .000001 inch) with a lay or graining direction parallel to the motion of cartridge insertion. This extra smooth surface finish eliminates unnecessary wear on the plastic-bodied tape cartridges. To help protect this smooth finish, the deck should be clear anodized to the MIL-A-8625 specification. Anodizing is an electrochemical plating process that hardens the surface of aluminum, gives increased corrosion resistance and preserves aluminum's original aesthetic appearance. As with all components, the deck must be closely inspected by the Quality Control department to guarantee 100% adherence to all design parameters.

Solenoid - The purpose of the solenoid (see Fig. 1) in a tape transport system is to supply motion to the drive cam and cross shaft to rotate the pressure roller into position against the tape and motor capstan. There are two major components of a solenoid, the body and plunger. The body contains the electromagnetic coil and should be of sturdy design with a non-corrosive zinc or cadmium plated finish. The body should have adequate means to aid in mechanical mounting of the solenoid. The plunger should be coated with a Teflon based low friction finish (Impreglon #218 or equal) to assist in repeated smooth operation. A tapped hole located in the face of the plunger is required to mechanically attach the cross shaft drive cable. Equally spaced holes around the circumference of the exposed end of the plunger will aid in rotating the plunger during final adjustment. The solenoid converts DC voltage into linear motion of the plunger by the creation of a magnetic field within the body of the solenoid. Solenoid performance is best described graphically (see Fig. 2) in terms of stroke/force/voltage curves. For any given solenoid, the force at the plunger varies with stroke position and applied

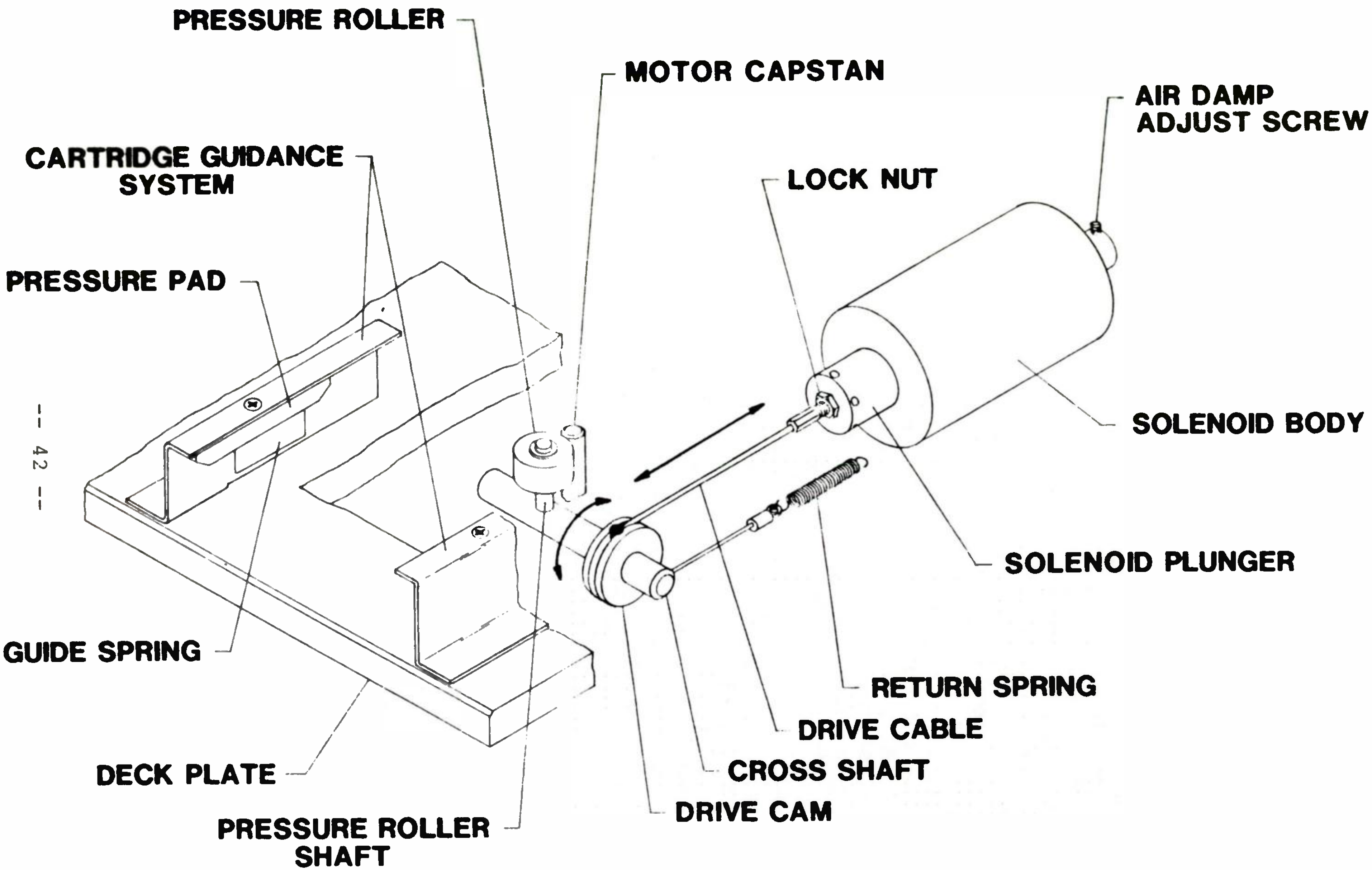

FIGURE 1: TAPE CARTRIDGE TRANSPORT SYSTEM

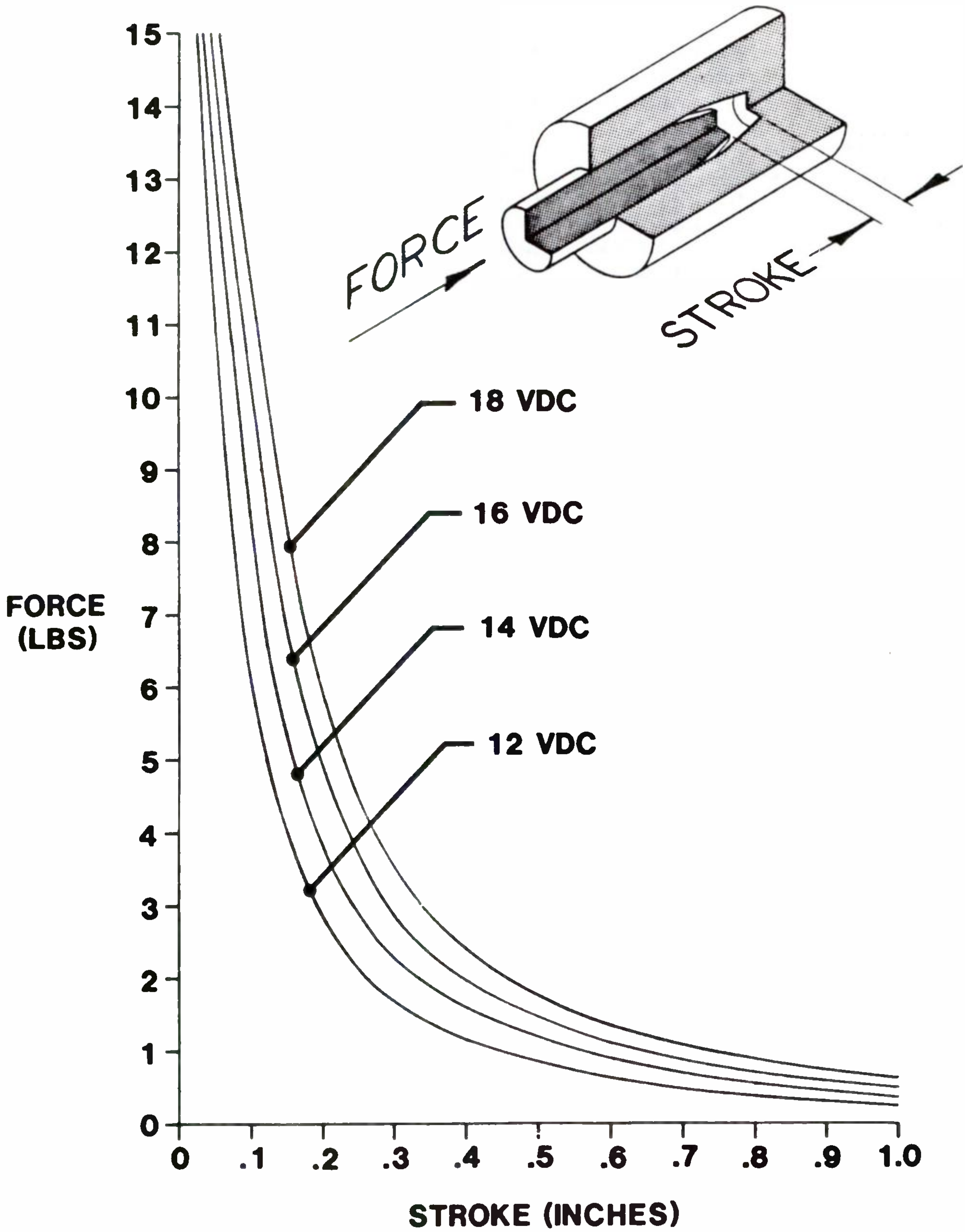

FIGURE 2: SOLENOID PERFORMANCE CURVE

voltage. The stroke measurement is defined as the plunger distance out from the fully seated or bottomed position. Internal or coil temperature can have a weakening effect on solenoid performance. As the coil temperature increases, so does the coil electrical resistance. This increased resistance will result in lower current flow for a constant voltage level. This decreased current will decrease the force performance of the solenoid. To overcome the effect, the use of a constant current supply to the solenoid and adequate thermal dissipation and ventilation are required.

Solenoids in tape drive systems must not only be strong pulling but also quiet in operation. To slow down the fast moving plunger, an adjustable air dampening device is mounted to the rear of the solenoid. Dampening adjustment of the solenoid allows the drive system to be adjusted within the NAB specifications for transport start and stop times and yet be quiet in operation.

Although stroke/force/voltage performance will vary from solenoid to solenoid, a tight quality control program with sample testing will prevent many problems.

Drive Cable - The element that transfers the linear force from the solenoid plunger to the drive cam is the cross shaft drive cable (see Fig. 1). This linkage must be strong, flexible, lightweight, adjustable and non-magnetic. The use of a .046" dia. 7 x 19 stainless steel cable with an attached threaded brass stud meets all of these requirements. The maximum impact tension force for this cable in actual use would be 27 lbs., well below the cables breaking strength of 270 lbs. (Factor of Safety 10:1). Cable of 7 x 19 construction (7 strands of 19 wires each) is used where flexibility and good resistance to wear is required. Successful life testing of this cable to over 2 million cycles indicates that fatigue failure is not a problem. The lightweight design of this cable at .006 #/ft. versus miniature chain link at .094 #/ft. is beneficial in faster start times due to lower mass inertia. Cable drive systems are also much quieter than the rattle of chain driven designs. The brass threaded portion located at the end of the cable allows mechanical attachment to the solenoid plunger and prevents the conduction of any magnetic flux to the rest of the system. At mid-span of the cable is a swaged brass ball or bead that will fit into a socket on the drive cam. This ball-socket arrangement allows the cable to grip the cam. At the other end of the cable is a lug that accommodates the return spring.

Drive Cam - The function of the cross shaft drive cam (see Fig. 1) is to convert the .80 inch linear movement of the solenoid plunger/drive cable to rotary motion of 105 degrees. The cam shape, which is basically a wheel with an offset center hole, allows the trade off of slightly more linear motion of the solenoid for increased leverage on the cross shaft. This mechanical advantage allows for more pressure roller indentation and improved tape pull.

The drive cam is manufactured from Delrin 500 thermo plastic. Delrin acetal resins are homopolymers that possess the needed characteristics of strength, rigidity, resilience and resistance to creep, high fatigue endurance, resistance to repeated impact, low static and dynamic friction, with resistance to most solvents and chemicals. Lightweight design and good cosmetics are also an added plus for Delrin material. As with the drive cable, the drive cam has been cycled over 2 million times with nearly undetectable amounts of wear.

Cross Shaft - The cross shaft (see Fig. 1) transmits the rotary motion and torque from the drive cam to the pressure roller shaft. This rotation of 105 degrees brings the pressure roller up into position to engage the tape and motor capstan.

Material selection is critical in the design of the cross shaft. Non-corrosive, non-magnetic Type 303 stainless steel rod that has been centerless ground and polished is the best choice. Grinding the diameter to a tolerance of +/- .0005" with a surface finish of 8 microinches is essential. The tight toleranced diameter will assure wobble-free fit into the equally precise hole in the deck plate. Addition of silicon grease lubrication in tandem with the extremely smooth polished finish will result in a low friction, non-binding operation.

Pressure Roller Shaft - The purpose of the pressure roller shaft (see Fig. 1) is to support the pressure roller on the cross shaft. This shaft must mechanically locate the pressure roller off the surface of the deck plate in accordance with NAB standards. These dimensions must be held precisely so the pressure roller will rotate up through the cartridge key hole and engage the tape properly.

Due to the high velocity rotation and resultant impact of the pressure roller against the motor capstan, the pressure roller shaft must be able to withstand high stress loading. Stainless steel is again the choice because of its high strength, good ductility and non-corrosive properties. Adequate care should be given in fabrication of this shaft to use generous radius in all machining to eliminate unwanted stress concentrations. The bearing area of the pressure roller shaft should be ground to a diametrical tolerance of +/- .00025" with a polished surface finish of 10 microinches. The purpose of this grind and polish is two fold: one, to produce a low friction interface with the pressure roller bearing; and two, provide a tight bearing fit to prevent unwanted movement and resultant tape speed fluctuations.

Other features in this design include wrench flats or hex body to facilitate installation or removal of the shaft. A snap ring on the top end of the shaft allows for quick and easy removal of the pressure roller if required.

Pressure Roller - The pressure roller (see Fig. 1) rotates up through the keyhole in the cartridge. It pinches the tape between itself and the rotating motor capstan which causes linear movement of the tape. During normal operation, the motor capstan drives the pressure roller and the pressure roller in turn pulls the tape. Tape slippage occurs when the force required to pull the tape from the cartridge exceeds the force created by the pressure roller and motor capstan. At this point, the pressure roller and tape cease moving even though the capstan may continue to rotate. Tape pull is directly related to pressure roller indentation against the capstan for a given pressure roller design. More discussion of this relationship will be presented later in this paper.

The pressure roller consists of three parts, the roller, the bushing and the bearing. The roller material is polyurethane which is a copolymer thermoplastic elastomer. Polyurethane is the ideal selection for the roller because of the following characteristics: outstanding wear, flex and tear resistance, high resilience, good elasticity, very slow aging and resistance to most solvents and chemicals. After casting or molding, the surface must be ground to a 32 microinch finish for good adhesion to the tape. Durometer readings, which measure the hardness of elastic products, must be tightly controlled. The outside diameter is held to +/- .003" to prevent unwanted elliptically shaped parts. The natural clear transparent color of urethane is selected to aid Quality Control inspection of any internal air pockets, voids or defects.

The urethane roller is permanently bonded to the aluminum bushing during the casting/molding process. The bushing is the interface between the roller and the bearing. The inside diameter of the bushing must be held with tight tolerances of +/- .0005" to provide a snug fit between the bushing and the bearing.

The bearing used in the pressure roller is a non-metallic, self-lubricating type fabricated from Turcite material. Turcite is a PTFE (Polytetrafluoroethylene) thermoplastic based product. Turcite bearings have excellent wear resistant properties, low friction, low creep, good pressure/velocity bearing performance and are impervious to most chemicals.. This material was selected because of its selflubricating and quiet running operation.

Pressure roller fabrication is very critical in the performance of the tape cartridge machine. An improperly balanced pressure roller will play havoc with tape speed specifications. To solve this problem, it is mandatory that the completely assembled pressure roller system (roller, bushing, bearing) be match machined to maintain precision concentricity (.002" TIR) between the roller outside diameter and the bearing inside diameter.

Motor Capstan - This paper will limit itself to just the mechanical aspects of the cart machine motor namely the motor capstan (see Fig. 1). Precision shaft diameter is critical to maintain constant tape speed. A diameter tolerance of +/- .0001" is required on the stainless steel non-magnetic shaft. This shaft must be sandblasted and then hard chromed to just the right finish for proper tape pull performance. Not only is the static diameter of the shaft critical, but the shaft when rotated cannot wobble by more than .00015 inches. Just as in the case of the pressure roller, this rotating trueness is important to good cart machine performance.

The position of the motor capstan relative to the deck plate must be as near perpendicular as possible. Specifications on the motor drawing should call out for shaft to mounting block angular dimensions of 90 degrees +/- .25 degrees. As discussed before, the flatness and parallelism of the deck plate is key to proper component assembly of the motor and deck.

Incoming quality control of the motor to sort out any damaged or out of specification motor capstans is essential to producing a quality product to the customer.

Cartridge Guidance System - The tape cartridge must be exactly positioned on the deck plate to result in top performance. This positioning is the function of the cartridge guidance system (see Fig. 1). As the cartridge is placed in the mouth of the cart machine, the cartridge must be positioned firmly downward on the deck and also sideways against the right hand cartridge guide. The insertion of the cartridge is of course stopped by the head box main frame. These three bench marks, namely the deck plate, righthand cartridge guide and head box main frame, are specifically located by NAB code and cannot be violated. Pressure pads supply the downward force and can be adjusted for optimum drag for NAB type cartridges. The spring on the left hand cartridge guide forces the tape carrier to the far right during insertion.

The aluminum material used in the manufacturing of the left and right hand cartridge guides is adequately strong and is black anodized (MIL-A-8625) for increased wear resistance and cosmetic appearance. The pressure pads are molded from black Delrin 500, a plastic material of low friction coefficient and high abrasion resistance (see drive cam material).

Head Box - See paper titled: "PHASE LOK V HEAD BOX ASSEMBLY: DESIGN, OPERATION, ALIGNMENT AND MAINTENANCE" By Richard L. Anderson, Broadcast Electronics, Quincy, IL.

SYSTEM OPERATION

The purpose of the tape cartridge transport system is to correctly guide an inserted NAB cartridge in to position, rotate the pressure roller up into place against the motor capstan and pull the magnetic tape across the heads located in the head box. After pressure roller rotation, the system must reach a point of equallibrium. This balanced condition can best be described by the use of a free body diagram (FBD) (see Fig. 3). As in every free body diagram, the summation of all forces and moments (torques) must equal zero. In the diagram of Fig. 3, only the total moments about the cross shaft (Mcs) will be of concern. Thus:

$$Mcs = (Fs)(L1) - (Frs)(L2) - (Fpr)(L3) = 0$$

Where: Fs = Force Of Solenoid (lbs.)
Frs = Force Of Return Spring (lbs.)
Fpr = Force On Pressure Roller (lbs.)
L1 = Distance From Fs To Cross Shaft Center Line (in.)
L2 = Distance From Frs To Cross Shaft Center Line (in.)
L3 = Distance From Fpr To Cross Shaft Center Line (in.)

Knowing Frs, L1, L2 and L3 for the "C" series design, one can solve for Fs and Fpr. Those results are presented in Fig. 4.

It is quite obvious that the amount of pressure roller force (Fpr) is directly related to solenoid force (Fs). As solenoid force increases, so does pressure roller force. As pressure roller force increases, the urethane roller is pressed tighter against the motor capstan. This force causes increased indention of the pressure roller. This force versus indentation is well defined for a given urethane pressure roller with fixed diameter, height and durometer (hardness). Likewise, as the indentation increases on the pressure roller, the ability to pull tape also increases. This tape pull force (Ftp), pressure roller force (Fpr) and roller indentation relationship can be reviewed in Fig. 5 with the actual numerical results in Fig. 4.

So far, only the mechanical aspects of the pressure roller indentation and tape pull have been discussed. In reality, an optimum indentation must be determined in regards to wow/flutter readings. With too little indentation (less than .010") tape slippage occurs, with too much indentation (greater than .020") an unwanted rotating lumping action occurs. Thus, the ideal compromise between good tape pull and low wow/flutter readings is

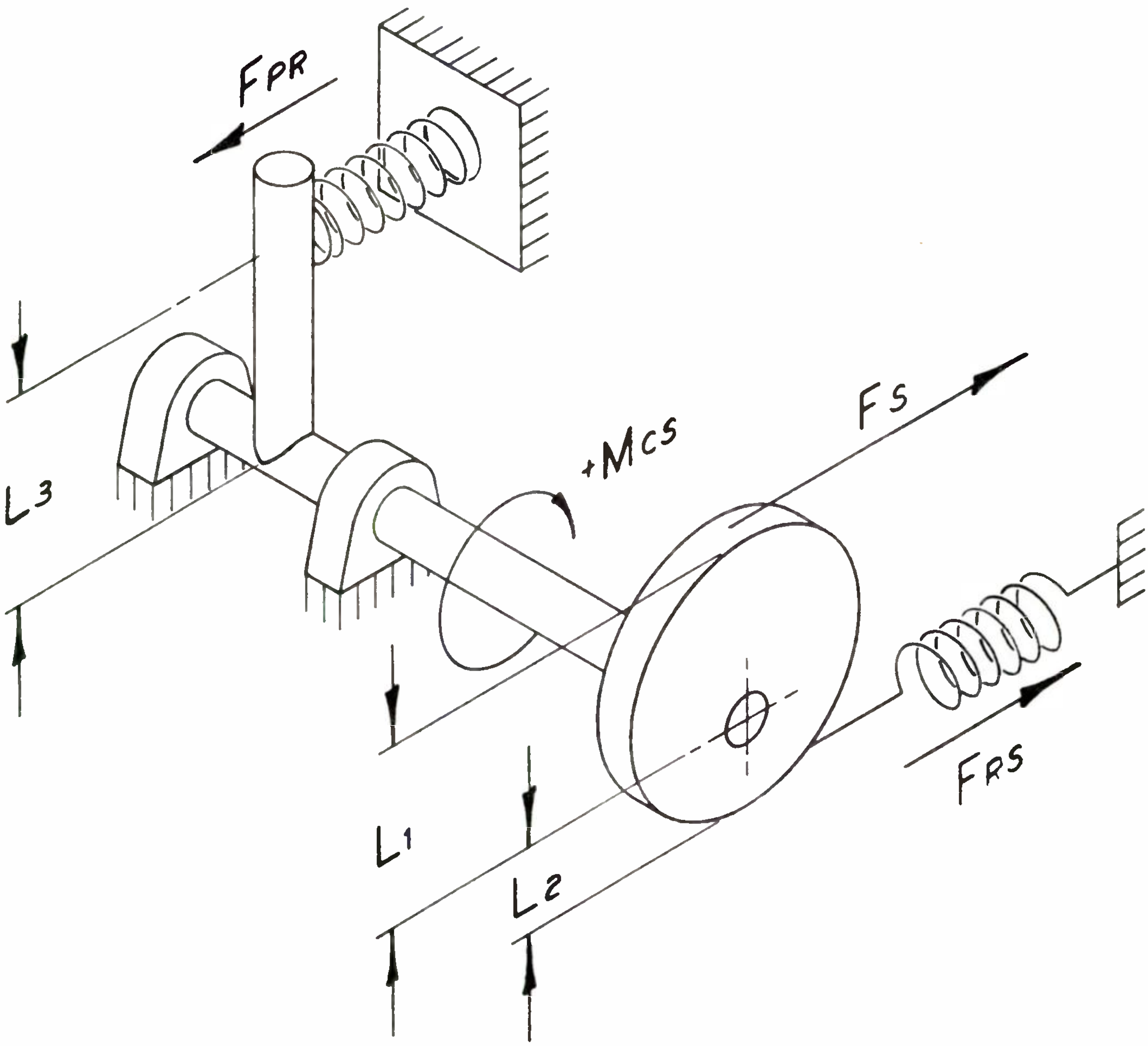

FIGURE 3: FREE BODY DIAGRAM (FBD) OF SYSTEM

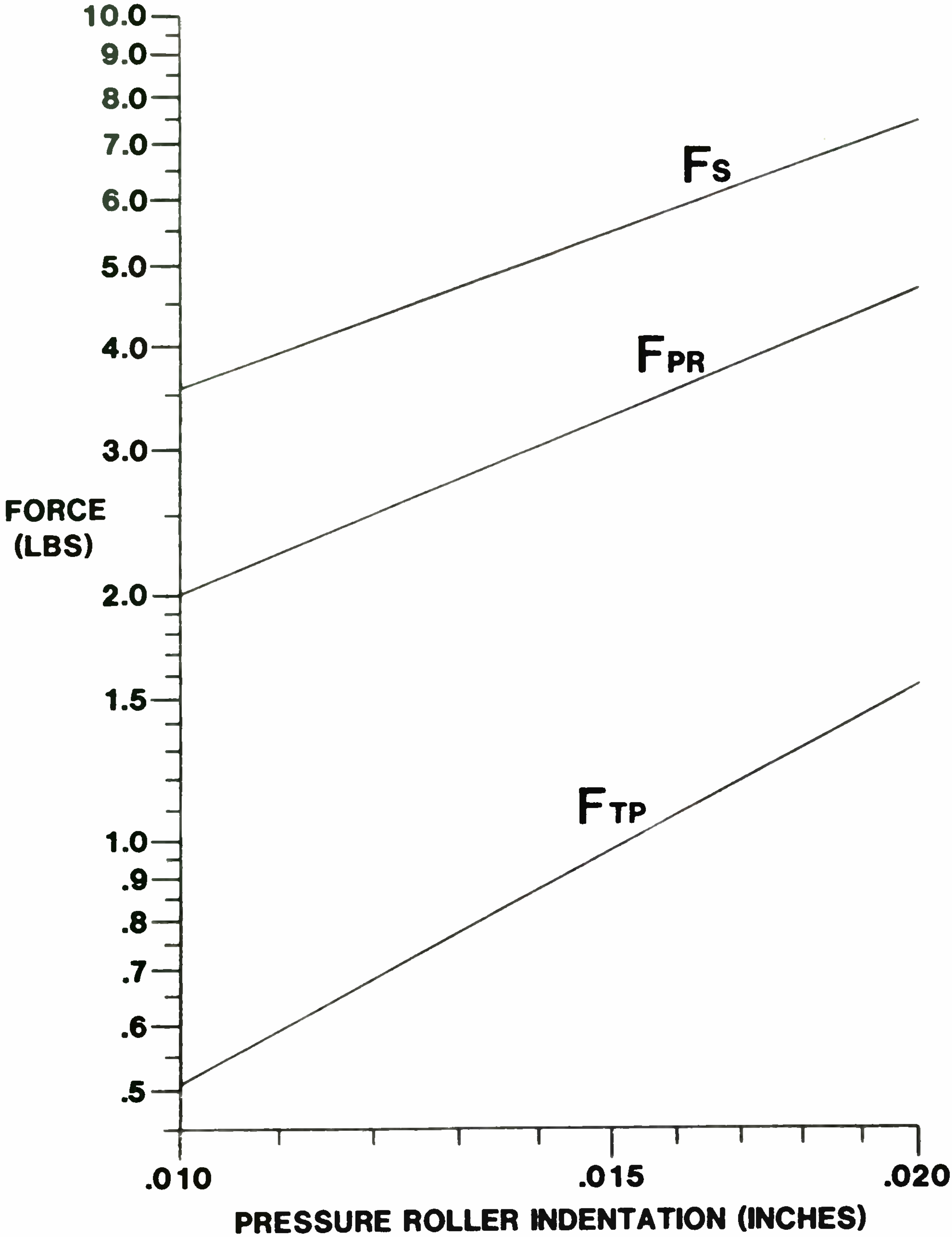

FIGURE 4: F_S, F_{PR} & F_{TP} VS ROLLER INDENTATION

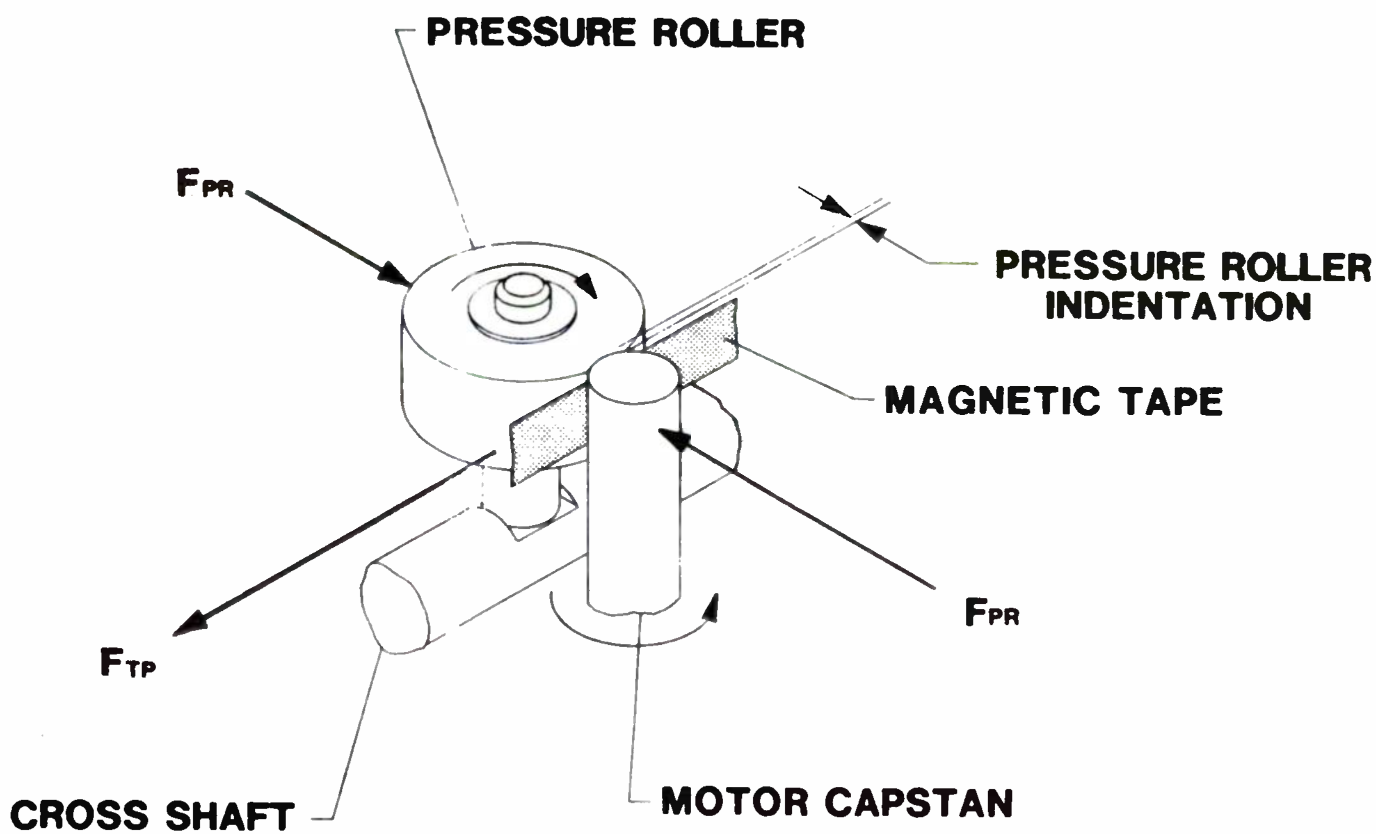

FIGURE 5: PRESSURE ROLLER/CAPSTAN DRIVE SYSTEM

near the .015" indentation point. Therefore, the tape transport system must be designed to stabilize at this point of equilibrium.

Adequate tape pull is a controversial topic for today's broadcaster. With the recent introduction of HOLN (high output, low noise) lubricated tape in NAB cartridges, the ability to successfully pull these tapes is in question. Due to these so called "hot" tape's smoother surface, lower coefficient of friction and tendency to pack tighter, it requires additional force to pull these tapes out of the cartridge. The "C" series tape transport system was especially designed to be strong pulling without affecting tape speed. At the indentation value of .015", a tape pull of 1.0 lbs (16 oz) is accomplished. From extensive testing of NAB cartridges with "hot" tapes, this 1.0 lb pull capability of Broadcast Electronic's system is comfortably sufficient.

SYSTEM MAINTENANCE

The three goals of maintaining any piece of equipment are: keep it clean, keep it oiled and keep it adjusted. These goals are quite appropriate for the tape transport system.

Scheduled housecleaning of the system should be standard procedure for every cart machine owner. The overall system should be kept free of dust, dirt and foreign contamination. The motor capstan and pressure roller should be cleaned daily with isopropyl alcohol to remove all traces of tape oxide or tape lubricant. Use of a clean cloth or Q-tip is recommended. Periodic cleaning of the inside diameter of the pressure roller bearing with isopropyl alcohol, although not required, is beneficial to increased bearing life.

Since all of the components needing lubrication are either factory greased or self-lubricating, the "C" series system is essentially maintenance free. For instance, the motor contains permanently lubricated ball bearings and the Turcite pressure roller bearing is self-lubricating. The cross shaft is factory lubricated with silicone based grease and the drive cable is pre-oiled. The teflon based coating on the solenoid plunger likewise requires no additional maintenance.

Proper adjustment of the tape transport system is essential to good, reliable performance. At times, periodic adjustment may become necessary for the cartridge guidance system, pressure roller/motor capstan and solenoid setting. Refer to the technical manual for instructions and gaging information.

CONCLUSIONS

Proper design, operation and maintenance of a tape cartridge transport system is critical to a successful broadcasting organization. The cart machine manufacturer must apply sound fundamental engineering principles to the design. Broad knowledge of mechanisms, statics, dynamic motion, material selection, and manufacturing technique are just a few of the specializations needed. Mixed in with all this theory must be a foundation of common sense and a commitment to a quality product.

The cart machine owner must understand the operation and maintenance requirements of his own unique system in order to produce a cost effective yet quality signal to the listener.

ACKNOWLEDGEMENTS

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AUTOMATIC PHASE CORRECTION FOR TAPE CARTRIDGE MACHINES

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Time delay (phasing) errors between the left and right channels in stereo tape cartridge machines causes erratic high frequency loss and other compatibility problems when the channels are "summed" to monaural either before transmission or within a monaural receiver tuned to a stereo broadcast. These problems are created by azimuth errors between the playback head gap and the material recorded on the tape. The common causes of azimuth errors are head misalignment, gap scatter and the time dependent variations of the tape cartridge and its related tape guidance system.

Presently used cartridge machine phase correction systems use a one time alignment during recording to correct for an average phase error value. This average correction only partially addresses the problem since the amount of correction needed changes as the tape cartridge mechanism revolves. Real time correction is required to completely correct phase errors. Stand alone systems are available to correct for phase errors, but they are expensive and require encoding of a reference signal on each tape.

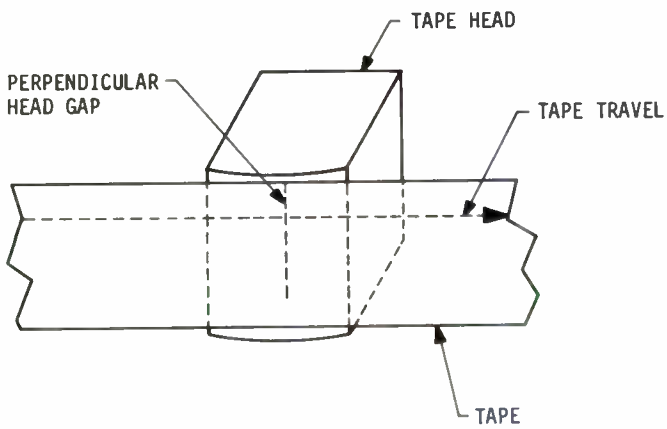
This paper describes a technology that corrects phase errors in real time during tape motion without the encoding of a reference signal.

Limits Of Performance

In order to set the design limits of the correction circuits, the performance limits of the tape cartridge medium need to be defined. The parameters that we are most interested in are azimuth error and its effects, signal-to-noise, frequency response, separation and distortion.

Azimuth

Azimuth refers to the head gap orientation with respect to the direction of tape travel. Absolute azimuth denotes that the head gap is perfectly perpendicular to the direction of tape travel as shown in Figure 1.



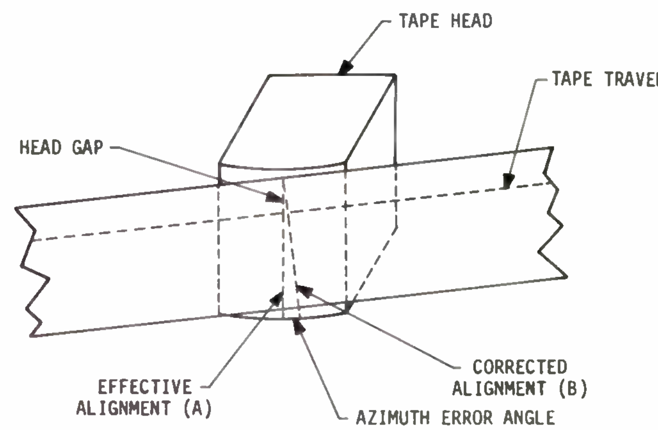


Figure 1. Absolute Azimuth

Figure 2. Relative Azimuth Error

Relative azimuth error depends on the difference between the azimuth of the material recorded on the tape and the azimuth of the playback head as shown in Figure 2. Relative azimuth errors cause two problems in tape cartridge machines. The first is a non-recoverable loss of high frequency information during playback. Figure 3 shows the amount of high frequency loss versus relative azimuth error according to the formula:

$$L=20 \log \sin (180\ T)/3.14159*T$$

L= loss in dB
T= $\tan (A)*F*W/V$
A= relative azimuth error (degrees)
F= the frequency of interest (Hz)
W= track width (inches)
V= tape speed (inches/second)

Azimuth errors also cause time delay (phase shift) errors between the two audio channels. This interchannel time delay can cause image shift in stereo systems and cancellation of some signal components in summed monophonic systems. The cancellation effect is most pronounced at high frequencies because the differential time delay results in a phase error which increases with frequency according to the formula:

$Pe=360*Td*Fs$

Fs is the frequency of interest.
Td is the interchannel time delay.
Pe is the interchannel phase error.

From Figure 3 it can be seen that an azimuth error of 0.5 degrees causes a 10.5 dB loss at 15 kHz. This is an interchannel phase error of 40 degrees at 1 kHz, which is equivalent to an interchannel time delay of 115 microseconds.

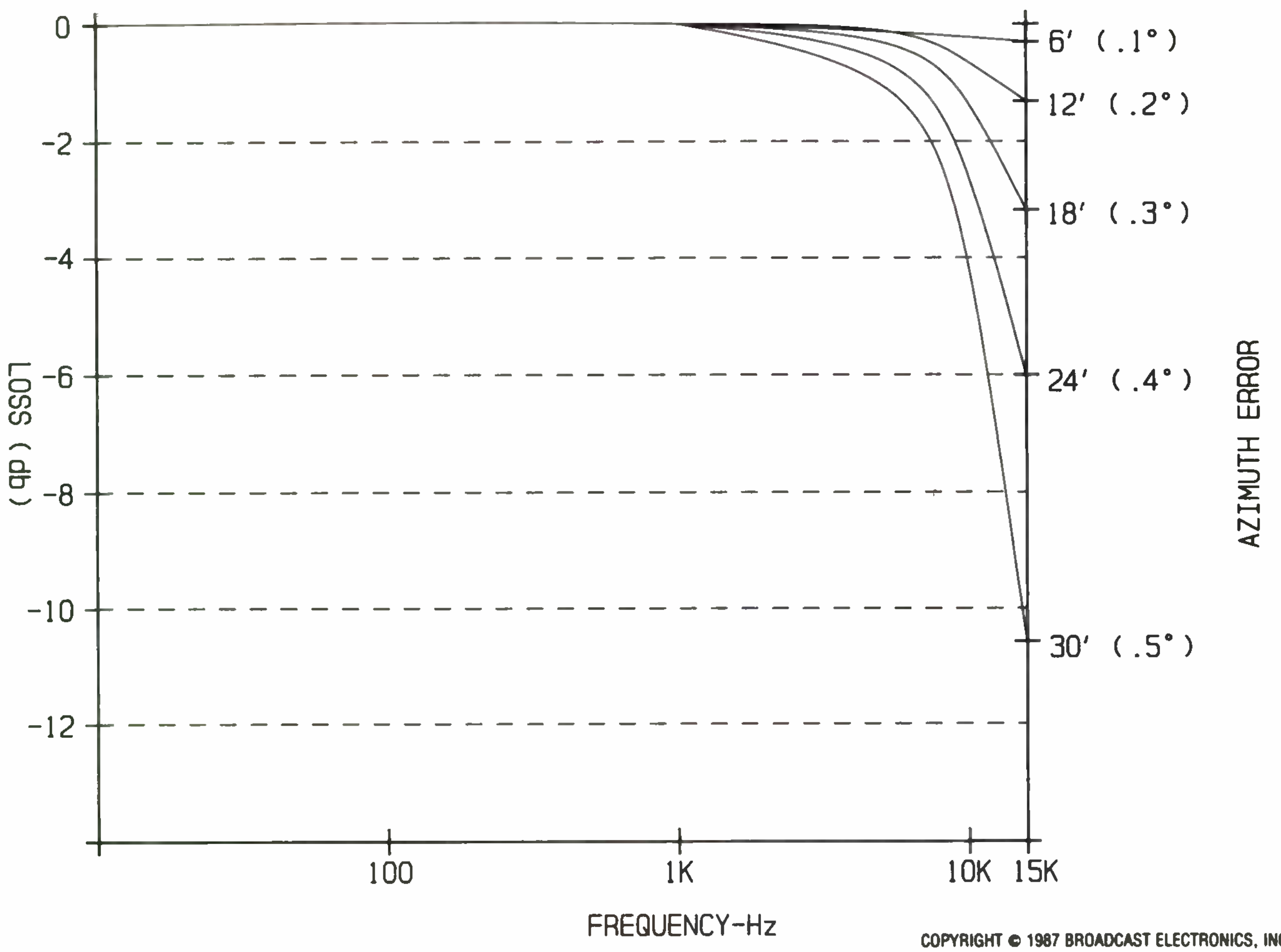


Figure 3. Loss as a Function of Frequency and Azimuth

This 115 microseconds of delay causes complete cancellation of 4.3 kHz and 13 kHz components in a mono sum signal as shown in Figure 4. The maximum tolerable system delay error would seem to be less than 16 microseconds. This is equivalent to 90 degree interchannel phase error at 15 kHz (relative azimuth error of 0.07 degrees) which gives a maximum stereo frequency response loss of 0.15 dB at 15 kHz. The maximum mono sum loss would then be 3 dB at 15 kHz.

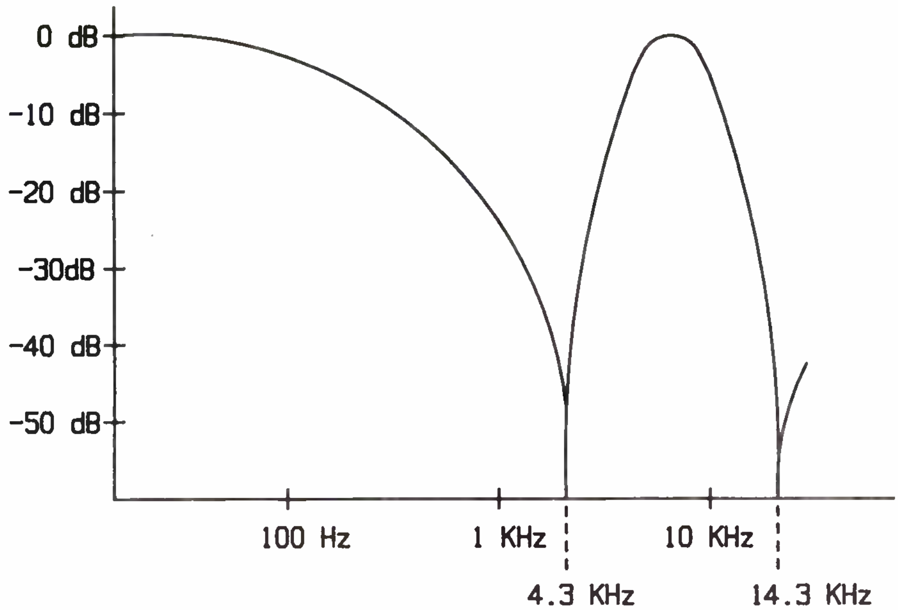


Figure 4. Summed Monophonic Frequency Response vs. Delay

It is obvious from these figures that the onset of monophonic signal degradation occurs with much smaller relative azimuth errors than those which affect the stereo channels only. This information on the amount of error necessary to degrade the monophonic versus the stereo program material sets the necessary time delay correction range. Along with the typical cartridge machine performance data discussed next, it sets the boundaries for the performance of the phase correction system.

Signal-To-Noise

The signal-to-noise of a tape machine is determined by the tape type and operating level. Without some form of noise reduction, an unweighted S/N of 60 dB below 250 nWb/m pulling tape is achievable at this time.

Frequency Response

The frequency response of a tape cartridge machine is limited by the amplitude of the low frequency contour effects of the head. Newer head designs provide a frequency response of ±1 dB from 30 Hz to 16.5 kHz.

Distortion

The distortion performance of a tape cartridge machine is limited by the tape type and the recorded level. Newer tape formulations allow distortion figures of under 1% at record levels of under 250 nWb/m.

Separation

The separation performance of any tape machine is dominated by the tape heads. While separation performance of 60 dB is possible at frequencies less than 1 kHz, inductive coupling between the coils of each channel limit the separation performance to the neighborhood of 35 dB at 16 kHz.

In order for the phase correction circuitry to be transparent to the cart machine performance, it needs to exceed the performance specifications given below in Table 1.

Phase Correction Performance Minimums

Distortion	$<0.5\%$
Signal-To-Noise	>70 dB
Frequency Response	$<\pm0.25$ dB 40 Hz-16 kHz
Separation	>60 dB 30 Hz-16 kHz
Correction Range	$>\pm115$ microseconds ($>\pm620$ degrees at 15 kHz)
Correction Error	$>\pm16$ microseconds ($>\pm90$ degrees at 15 kHz) ($>\pm30$ degrees at 5 kHz)

Mono Compatibility

As was detailed above, the onset of monophonic signal degradation occurs with much smaller relative azimuth errors than those which affect the stereo channels only. Monophonic listeners still form the largest group of listeners for all broadcasters. According to industry standards groups, over 90% of AM and TV listeners and 50% of FM listeners regularly listen in mono.

Compared to the smooth frequency response loss of the stereo channels with increasing azimuth error, the degradation of the summed monophonic signal takes on a particularly offensive quality. For example, an inter-channel time delay error of 50 microseconds totally cancels any 10 kHz response in the summed signal. The response loss at 5 kHz is only -3 dB and the signal amplitude also rises again above 10 kHz as shown in Figure 5.

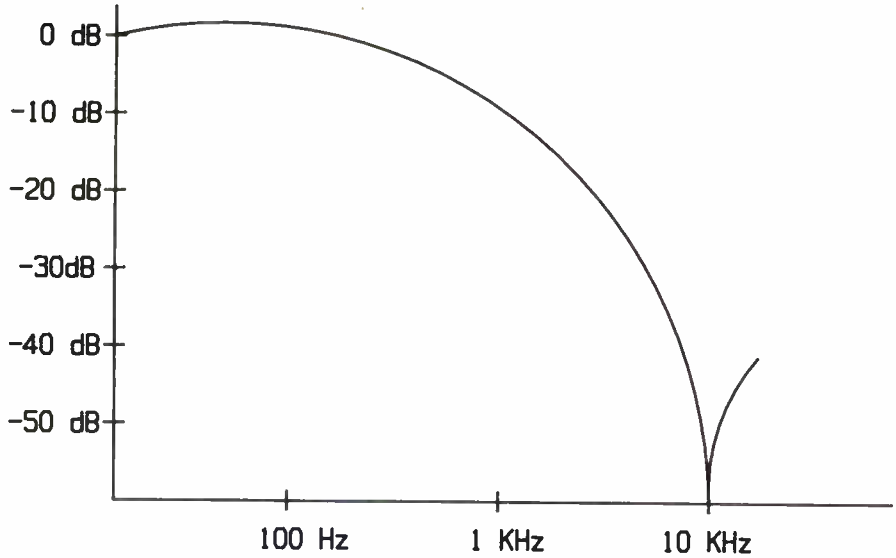


Figure 5. Summed Monophonic Response vs. Frequency (50 uS)

The objectionable nature of this degradation is due to the fact that this mid-frequency notch is not normally present in naturally occurring sound sources.

The time dependent skew of the typical tape cartridge magnifies the effect of this notch by shifting the notch back and forth in frequency at a slow rate dependent on the speed of the tape, the cartridge machine positioning system and the cartridge construction. The notch then audibly "swishes" up and down in frequency as the tape is played. These considerations require a real time phase correction system to remove all annoying artifacts from the monophonic signal.

Methods of Phase Correction

There are several available methods to correct phase in cartridge systems. They tend to fall into two broad categories: encoding systems and non-encoding systems.

Encoding Systems

Encoding phase correction systems usually inject some type of control signal onto the tape when it is recorded to define the correct phase relationship of the right and left channels. The advantage of the encoding systems is that the correction circuitry can be optimized for a known reference signal. For example, one system uses a modulated 19 kHz pilot signal recorded on both audio channels. Another system records left channel audio on the cue track for a reference.

While not usually thought of in these terms, a system which uses sum and difference matrixing techniques is also an encoded system. This process uses one channel for sum (L+R) information and the other channel for difference (L-R) information. Each playback machine must have a decoder (dematrix) and each recorder must have an encoder (matrix).

When they are actually used, encoded systems usually give the best phase correction performance. The disadvantage of encoded systems is the need to encode each cartridge in the system to take advantage of the correction capability. In order to actually use the correction performance of an encoding system, the entire cartridge library, all agency spots and any other cart not recorded with the encoding system must be re-recorded. Encoding systems also require each playback cartridge machine to have a decoder assigned to it, which imposes large cost, maintenance and complexity penalties on the entire audio system.

A subset of the encoding phase correction system is a cart machine system that mechanically adjusts the relative azimuth of the record head to that of the playback head during a setup procedure. This allows the machine to correct for the average phase error of that one tape machine/ tape cartridge system. This system ignores the questions of machine to machine interchangeability, cartridges that were not recorded on that system (eg. agency spots) and changes in tape cartridge phase performance due to wear and minor damage. It cannot correct for real time changes in phase error due to the rotation of the cartridge mechanism, which creates time dependent variations in the amount of tape skew.

Matrix encoding systems do not solve any phase problem, they just introduce the problem in another form. With phase problems in the discrete audio channels, there will be phase problems in the matrixed audio. In matrix form the result is usually poor separation and a poor stereo image for the stereo listener. To insure good separation, the amplitude and phase characteristics of the L+R and L-R channels must be tightly controlled. Most tape head amplitude balance specifications are on the order of ±3 dB and phase dispersion is rarely specified. Figure 6 gives the resulting separation if amplitude and phase errors between the L+R channel and the L-R channels are known.

Since the channel signal-to-noise ratio for a tape machine is fixed, it is also likely that the final discrete signal-to-noise ratio will degrade. Most stereo signals have much more L+R than L-R information. In the worst case (L or R only), the L+R channel will be 6 dB lower than a discrete channel. This is because when L=R the amplitude is twice the single channel value (6 dB). When the decoded noise contribution of the L-R channel is added during dematrix, there can be as much as a 9 dB degradation of signal-to-noise.

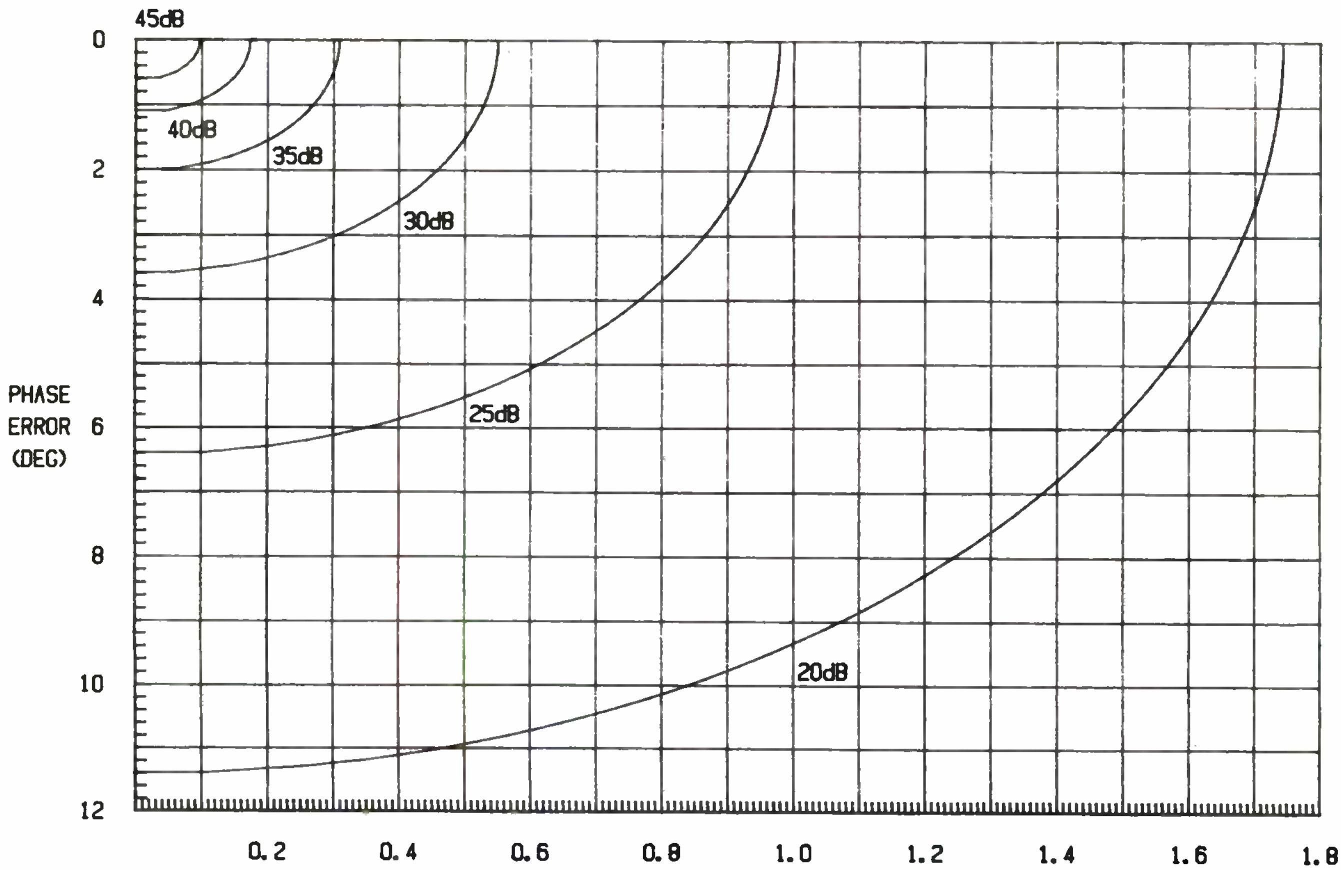


Figure 6. Separation vs. Differential Amplitude and Phase

Non-Encoding Systems

Non-encoding phase correction systems are based on the fact that stereo program material has a considerable amount of monophonic content and that this monophonic content can be used to guide the correction process. As was shown in the section on azimuth effects, degradation of the monophonic content of the stereo program occurs well before any degradation of the actual stereo information.

In order to use the monophonic content of a stereo program to correct for time delay (phase) errors, it is necessary to find a way to extract the time delay information from the audio signals on the tape. Signal theory points out a way by using a valuable property of signals called the autocorrelation function.

The autocorrelation function is a time function of the signal. It indicates the degree to which the signal is related to values of itself in time. This function is obtained by multiplying the current signal by a delayed sample of itself and averaging over the sample time. At zero delay, the signal is multiplied by itself and the value is the signal power. As the amount of system delay is increased the value of the autocorrelation will decrease. The autocorrelation function will always have its largest amplitude at zero delay, except for sine waves which have additional, equal value peaks at multiples of the period.

If a signal is multiplied by a delayed sample of another signal, the result is the cross-correlation function, which represents the amount of common information in the signals. Using the two channels of a stereo system, the cross-correlation will then represent the autocorrelation of the monophonic components. This is the information needed to extract the time delay information from the stereo signals. Using a servo system to maximize the value of the correlation function by linearly delaying the leading channel, will allow real time correction for interchannel time delays.

Unfortunately, finding the peak of the correlation function requires the evaluation of the function at several points and a search to find the largest peak. This process does not lend itself to cost effective real-time implementation in a servo system. However, knowledge of the performance limitations of the system to be corrected allows the design of real-time signal processing techniques to simplify the evaluation of the correlation function. Because each channel is processed before the correlation, the system design can be limited to detecting the zero point of the function, the central concept of any servo design.

This design allows a low cost, high performance real time phase correction to be designed for almost any stereo system. The design of the signal processing circuitry will be dependent upon the performance characteristics of the stereo source device. Since the optimum signal processing circuitry for each source device type is different, there is no easy way to use this technique to develop a "universal" phase corrector. Each type of stereo source device would require different signal processing circuits. Using one signal processing device would require re-adjustment for each source. If a compromise setting is attempted, degradation of both monophonic and stereo performance would result.

System Explanation and Performance

System Explanation

A block diagram of the system is shown in Figure 7.

The right and left audio channels are low pass filtered, then input to linear audio delay lines. The audio delay lines were picked as appropriate technology for a cartridge machine corrector system because they have a linear time delay versus frequency characteristic. It is very difficult to build analog phase delays that have a linear delay versus time characteristic. The delay line audio performance, while not up to compact disc audio quality, is more than adequate for tape machine use. The harmonic distortion is less than 0.25% and the signal-to-noise is greater than 75 dB below normal signal level. The left delay is fixed at one millisecond, the right delay is variable from 0.8 milliseconds to 1.2 milliseconds. This gives a maximum delay range of ±200 microseconds (±1080 degrees at 15 kHz).

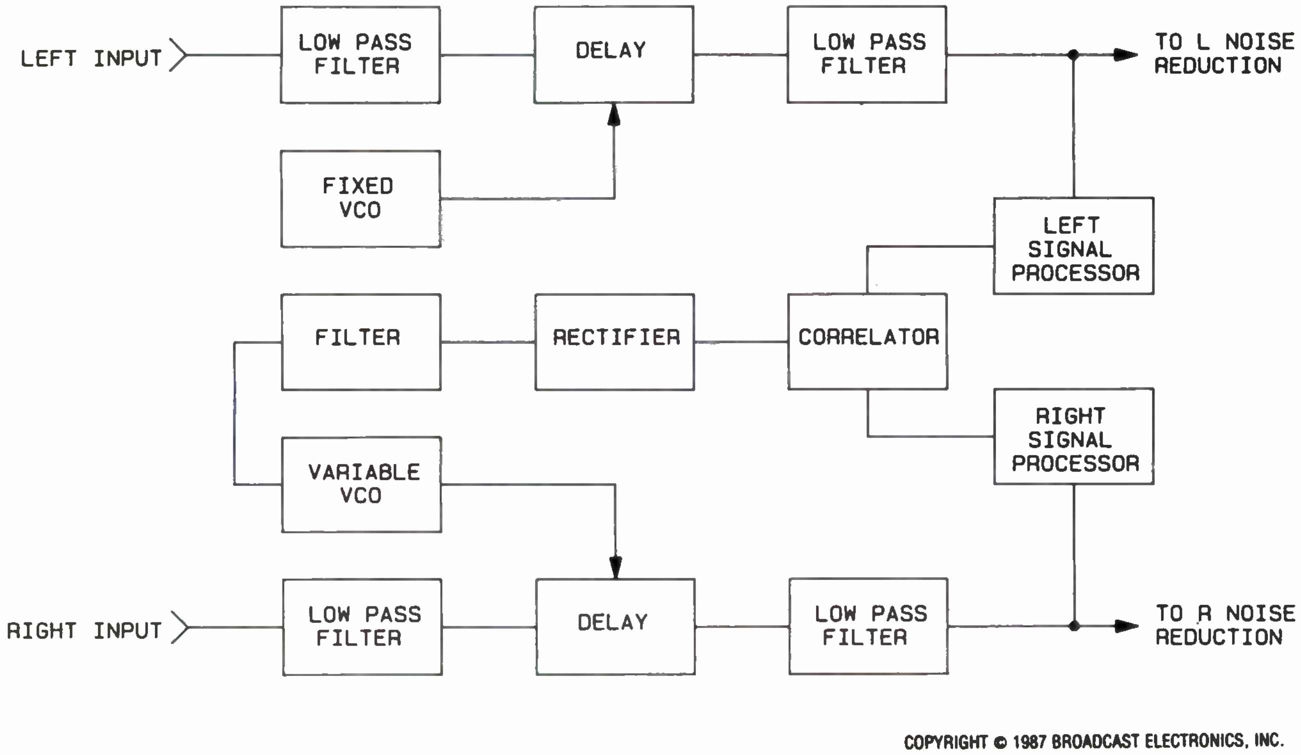
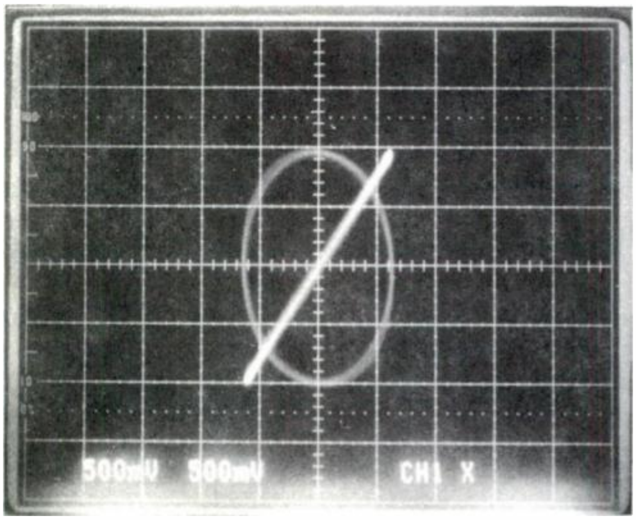


Figure 7. Tape Cartridge Phase Correction System Block Diagram

After the delays, the audio is low-pass filtered to remove sampling artifacts and is passed on to the noise reduction circuitry. The audio is also sampled at this point for the phase detector signal processor. After signal processing the signals are correlated, rectified and filtered. This filtered, level shifted signal is used to run the right delay clock, completing the phase tracking servo loop.

System Performance

The Lissajous figure shows before and after correction results for a single test tone of 7.5 Hz in Figure 8. The error is slightly greater than 90 degrees (oval trace) and the correction is better than 3 degrees (45 degree line). Figure 9 shows a before-and-after correction results for a pink noise test tape. These pictures were created by manually misaligning the playback head of the machine and then enabling the phase correction.



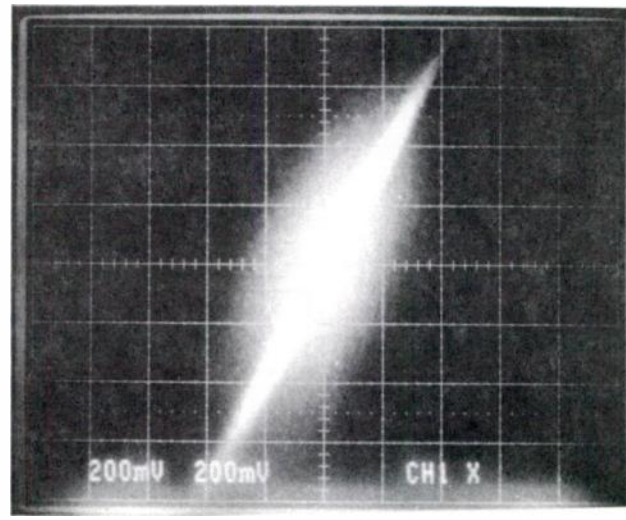


Figure 8. Tone-Phase Error Before and After Correction

Figure 9. Pink Noise Phase Error Before and After Correction

The trace in Figure 10 is a five minute sample of relative uncorrected cartridge machine playback phase error (degrees) versus time (minutes). A new 6 minute cartridge was used. It was recorded with a 7.5 kHz tone on the same cartridge machine used for playback. The large excursion in-phase at about 4 minutes is the tape splice. The trace in Figure 11 is a five minute sample of the same machine and tape as Figure 10, but with the phase correction in circuit. Without phase correction in circuit, the maximum peak phase error (excluding the splice) is 17 degrees, with short term variances of as much as 15 degrees. With the phase correction in circuit, the maximum peak phase error (excluding splice) is 1 degree, with short term variations of 2 degrees.

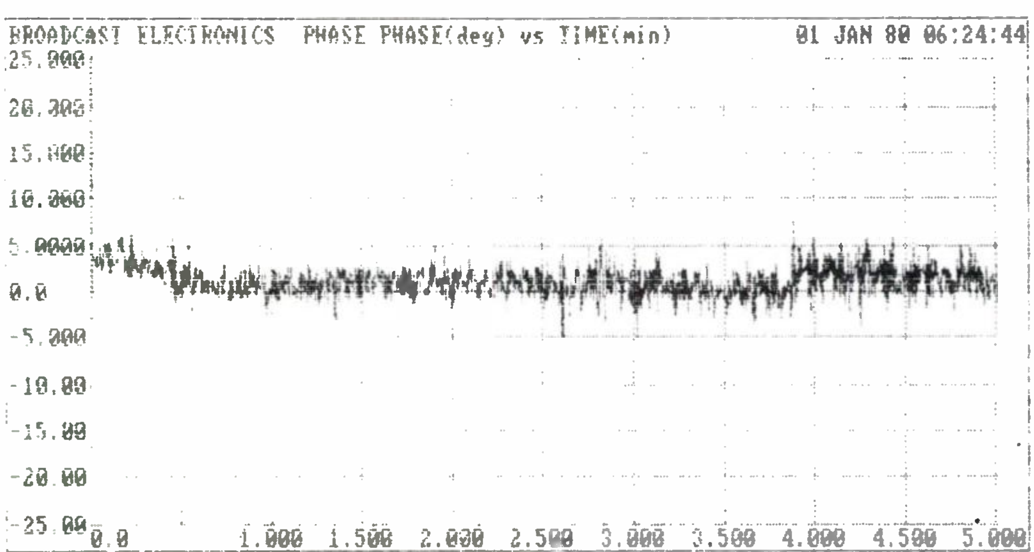


Figure 10. Uncorrected Phase Error Versus Time

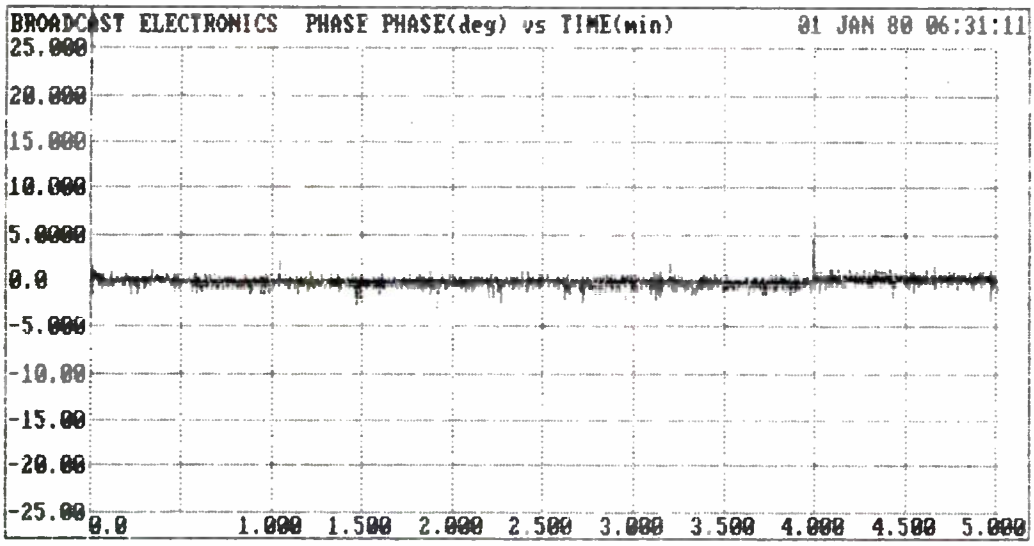


Figure 11. Corrected Phase Error Versus Time

This is an obviously an optimum system. It has a brand new cartridge and freshly tweaked record and playback alignment. Figure 12 shows a non-encoded five minute sample of the same machine, but with a different tape that was recorded on a different machine. The phase offset is 80 degrees. The maximum peak phase error is 85 degrees, with short term variations of 10 degrees. With the phase correction in circuit (Figure 13), the phase offset is eliminated. The short term error is reduced to less than 3 degrees.

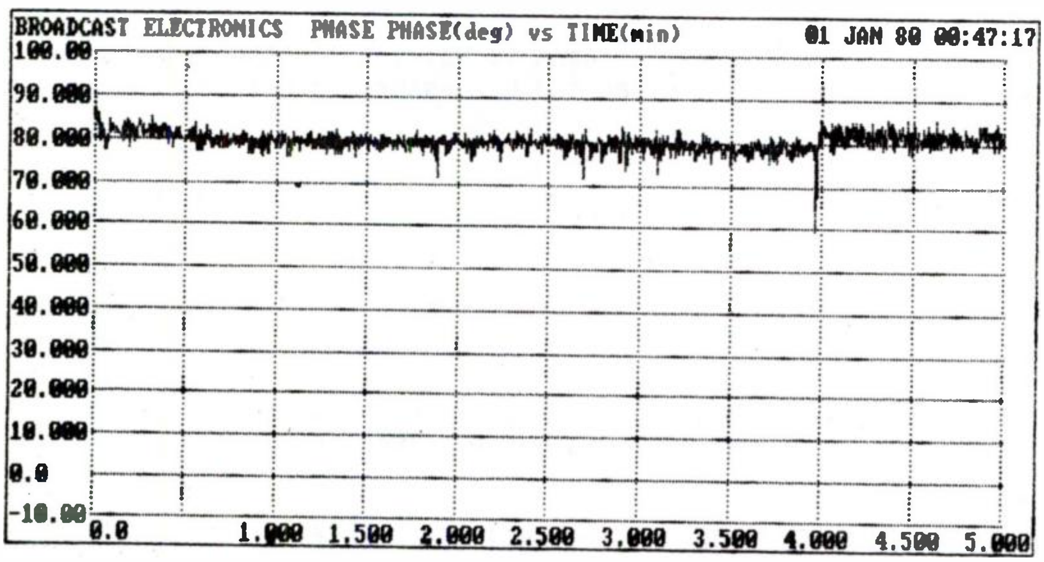


Figure 12. Old Cart-Uncorrected Phase Error Versus Time

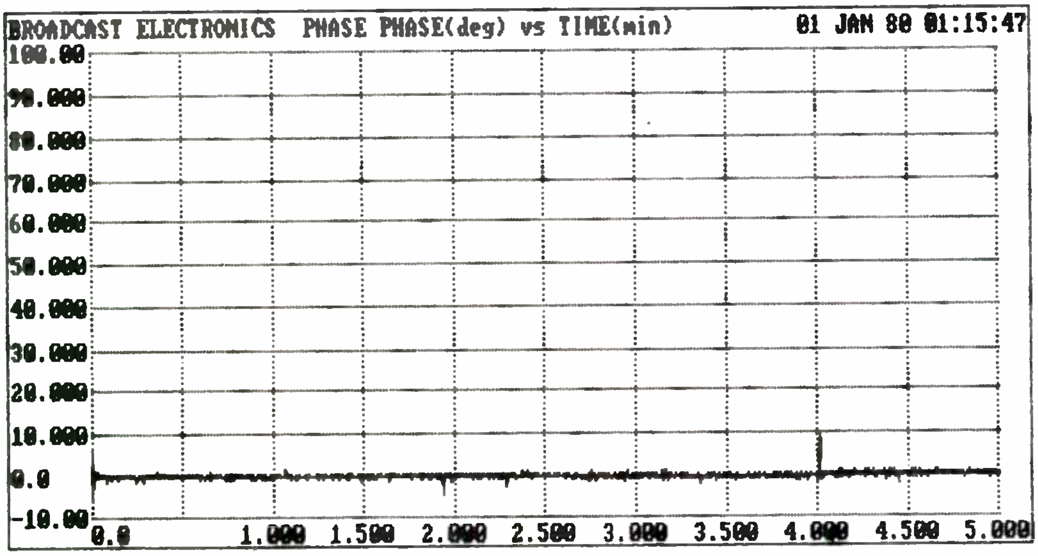


Figure 13. Old Cart-Corrected Phase Error Versus Time

Conclusion

A built-in, non-encoding interchannel phase correction system has been profiled in this paper. This cost effective, operator transparent system eliminates the monophonic compatibility problem for any tape cartridge machines playing any tape, without the inconvenience and expense involved with encoded systems.

Acknowledgements

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References

Burstein, Herman, "How Important is Tape Azimuth", Audio VOL.68, No.9, pp. 40-746, 1984.

Moris, A.H., Mullen, J.T., "Phase Error In Tape Cartridges for Radio Broadcast Service", Journal Audio Engineering Society, VOL.31, No.1/2, 1983.

Heinrich, Mer, "Delay-Lock Tracking of Stochastic Signals", IEEE Transactions on Communications, VOL. COM 24, No.3, 1976.

Cabot, R.C., Pavlok, R., "A High Accuracy Analog Cross Correlator", AES Preprint 1362, 60th Convention, 1978.

FUNDAMENTALS OF DIGITAL AUDIO

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Digital audio has been creeping into radio in various forms over the past few years. The emergence of the compact disc has presented an affordable media with adequate storage capacity and the promise of impeccable fidelity. Manufacturers of audio equipment have been struggling for the past few years to create effective digital audio hardware specifically for the broadcaster. As this equipment becomes a reality and finds its way into the studio, the station engineer will be faced with the task of specifying these new boxes, and maintaining them. An overview of the basics of digital audio (without the mathematical details) may help to shed some light on the difficulties being encountered by the manufacturers, and the new frontier that awaits the station engineer.

WHAT IS DIGITAL AUDIO?

Before engaging in the details of digital audio, one has to ask the question: why digital audio? The most common answer will be 'for its better sound quality.' A better answer would be 'for its ability to be stored and retrieved without degredation.' The digital audio recording process essentially converts audio to digital data, stores it, and converts it back to audio upon playback. Therefore, if the digital data can be retrieved accurately, audio quality is limited only by the conversion process. Analog recording contributes minute changes to the original audio information in the form of distortion and noise, and each time we re-record information it loses a little more of its original character.

In the digital recording domain, only the data representing the audio is involved in the recording process in the form of 1's and 0's. Recovering the data only amounts to differentiating between the two states, and the minute changes contributed by the recording process don't affect the audio signal. For years computers have been storing information in the form of digital data on various sorts of media which were subject to the same imperfections we are experiencing today. Being quite intolerant of lost data, the industry developed error correction codes which, before recording, would mathmetically encode the data in such a fashion that if an error occured, or if data were lost, the original could be reconstructed from the remaining data, provided the errors were not too large. Present day codes are virtually media independant; the Reed-Solomon code can be found applied to Winchester discs, magnetic tape and the compact disc. As more and more data is being crammed into smaller spaces, minute imperfections in media contribute to larger data losses, requiring both better media and more powerful error correction schemes. Both the computer data industry and the digital audio industry are engaged in new developments in these areas. The important issue here is that with digital audio as with computer data we can still exactly reproduce the information we store on the media.

THE CONVERSION

Converting the audio to a data format takes place in the analog to digital converter. This process is similar to that of recording a motion picture. Snapshots, called samples, are taken of the analog signal at continuous intervals in time. Each of these samples is assigned a number representative of the analog voltage at that period in time, and this number is recorded as data (Figure 1).

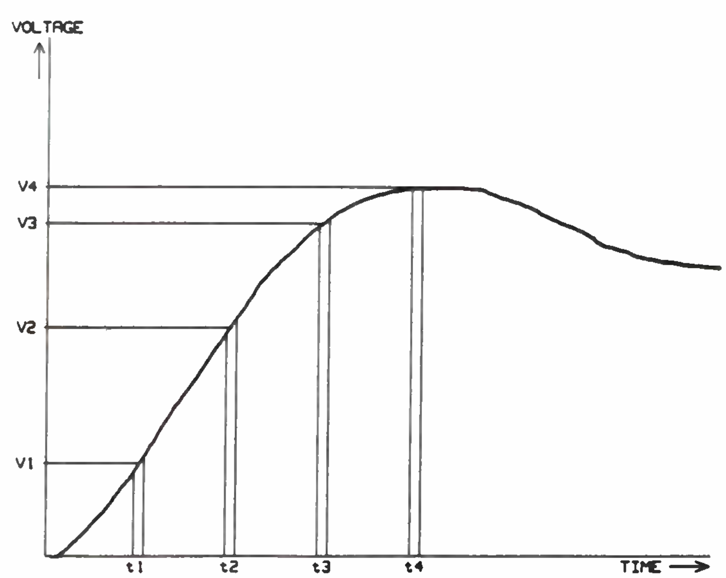


Figure 1 Discrete time sampling of an analog signal, creating a series of voltages to be recorded as data words.

Thus, we have represented the audio signal as a series of incremental voltages measured at discrete intervals in time. Playback then consists of putting each of these samples back in order at the proper time, much the same as playing back the many frames of a recorded motion picture. This process takes place in the digital to analog converter, where the data words are converted back to voltages, and the resultant waveform is smoothed out with a low pass filter. Although we are recording only a portion of the original analog signal, no information is lost if our sampling frequency is sufficiently high. More on this later. Professional digital audio systems use a sampling frequency of 48,000 samples per second; the compact disc uses 44,100.

AUDIO QUALITY

Our quest is for perfect audio. While the actual audio quality of a system is truly "in the ears of the beholder", there are certain measurable specifications developed for audio systems from which we have established some minimum standards for what we term "quality" audio. We use these numbers to compare various audio systems, and will find them to be effected quite differently when comparing analog and digital systems.

Dynamic Range or Signal to Noise

For the purpose of this discussion we will consider only dynamic range measurements, since dynamic range is essentially S/N measured from the largest signal value to the noise floor. Dynamic range in a digital audio system is directly proportional to the number of bits in the data word used to define the voltage level in each sample. These bits will then be recorded as data on our storage media. Each additional bit in the data word doubles the resolution of the analog signal. If, for example, the analog voltages ranged from 0.00 volts to 1.00 volt, a 1-bit system could resolve all input voltages into two groups. All voltages between 0.00 V and 0.49 V would be assigned the binary number '0' and voltages between 0.50 V and 1.00 V would be assigned a binary '1'. A two bit system has four levels of resolution (00, 01, 10, 11), while a 3-bit system has eight levels (000, 001, 010, 011, 100, 101, 110, 111) and so on (Figure 2).

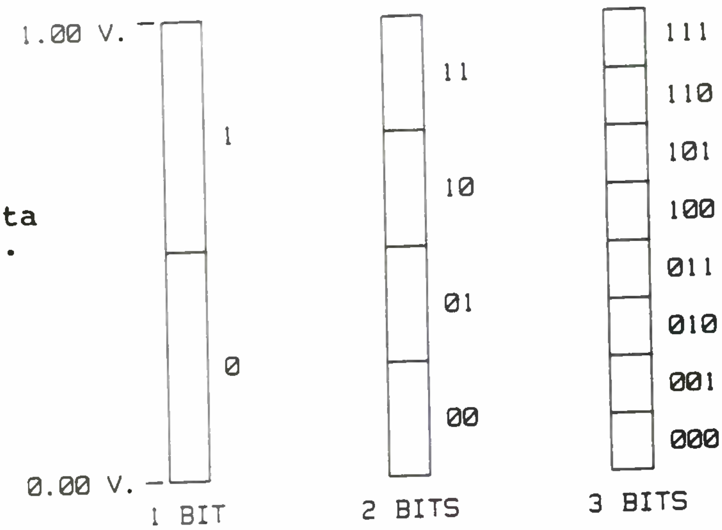


Figure 2 Resolution of a 1.00 volt signal into data words of 1, 2 and 3 bits.

The 16 bit system has thus far become the de facto standard in the digital audio system of storage. With 16 bits we can resolve each audio sample into 65,536 levels ($2^{16} = 65,536$). The mathematics tell us that we can accomplish 6 dB of dynamic range per bit of resolution.

Dynamic Range = (6 X N + 1.76) dB, where N is the number of bits.

A 16 bit system can therefore have a theoretical dynamic range of 97.76 dB.

The process of dividing the analog signal into discrete levels and assigning each a digital number is called "quantizing" and takes place in the analog to digital converter. We stated earlier that the sampling process, if done properly, causes no loss of information. This is not the case for the quantization process. For example, if the maximum voltage for a 16 bit converter is chosen as 10.0000 volts, then it would be assigned the value of 65,536 and each interval would represent .0001523 volts (Figure 3). Then, any voltage between 4.88281 and 4.88296 would be assigned a unique word of 32,000, or binary number 01111101 00000000, and this would be recorded for the sample. Generally the voltage at the center of the range (in this case 4.88289) would be assigned to the above 16 bit binary number, and all other voltages in that range would be in error. This error is referred to as "quantization error" or "quantization noise" since it ultimately manifests itself as analog noise (Figure 3).

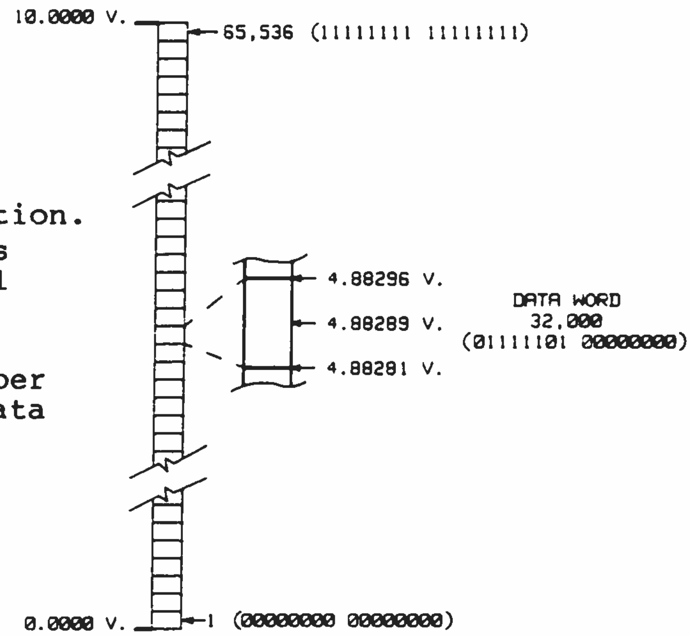


Figure 3 16 bit quantization. The 10.0000 volt range is divided into 65,536 equal parts of 0.000152 volts. Each part is assigned a unique 16 bit binary number which is stored as the data word for the sample.

Distortion

Distortion in a digital system is the result of inaccuracies in the quantization process, and therefore related to the number of bits in our system. The more bits we use to define the input voltage, the smaller the probable errors which show up as distortion components. These distortion components remain fairly constant over the range of input values, so unlike analog recording where the system distortion gets higher with increased recording level, the distortion numbers are better with higher input levels in digital recording.

Frequency Response

The frequency response of a digital system is limited by the sampling rate of the A to D converter and the associated filters used. The Nyquist criteria says that any sampled continuous waveform can be reproduced, provided the sampling rate is at least twice the highest frequency contained in the waveform. To achieve a frequency response with an upper frequency of 20 kHz in our digital audio system, we would need a sampling frequency of at least 40 kHz. The mathematical proof of this is quite complex. Consider that we are working with a band- limited signal which restricts the rate of change of the signal to a predictable maximum in any given timeframe. For instance, how often do we have to check our speedometer when driving down the road to maintain a constant speed of 55 MPH? If we checked every couple of minutes our speed could vary considerably. If we checked every 10 seconds, chances are our speed will stay pretty close to 55. The speed of the car is essentially low pass filtered by inertia and friction, making speed excursions predictable. In fact, knowing the nature of the low pass filter on our signal, we can then compute the missing data between samples. When performed properly, the sampling process does not destroy the original signal.

The sampling process itself can, however, create an entire set of unwanted frequencies if the input waveform is not properly managed. Any incoming signal to the sampler which is greater than half the sampling frequency will generate a frequency at the output which is the difference between twice the incoming frequency and the sampling frequency. In our example 48 kHz system, a 25 kHz input to the sampler will produce an output of 2 kHz (2 X 25kHz - 48kHz = 2kHz). We are all familiar with this phenomenon. Going back to my example of how this process is similar to that of making a motion picture, remember in the old westerns, when the outlaws were chasing the stagecoach, how the wheels always seemed to be going backwards? In this situation the frequency of the spokes in the wheels was higher than half the frame rate of the film. This phenomenon is called 'aliasing' and is the reason for the anti-aliasing filter in the analog to digital converter. This filter's job is to remove all frequencies that are greater than half the sampling frequency before they reach the sampling circuitry. This is no simple process, in that the filter must not affect our 20 kHz input signal, and yet must remove all 24 kHz and higher inputs. Typical in this application are 'elliptical' filters. An eleventh-order elliptical filter can be within 0.1 dB at 20 kHz, and 40 dB down at 22 kHz (Figure 4).

A similar filter resides at the output of the digital to analog converter (called anti-image). During playback, the data words are converted back to voltages which undergo a "sample and hold" process to bridge the time between voltage samples. This sampling produces a sideband frequency spectrum at the sampling frequency which is an image of the audio spectrum. Since our audio spectrum is limited to approximately 20 kHz by the anti-aliasing filter in the A to D converter, we will have generated spectral components in the area of 28 kHz (48 kHz -20 kHz). Again, a very sharp cutoff filter is required to remove these frequencies and not effect our audio frequencies at 20 kHz.

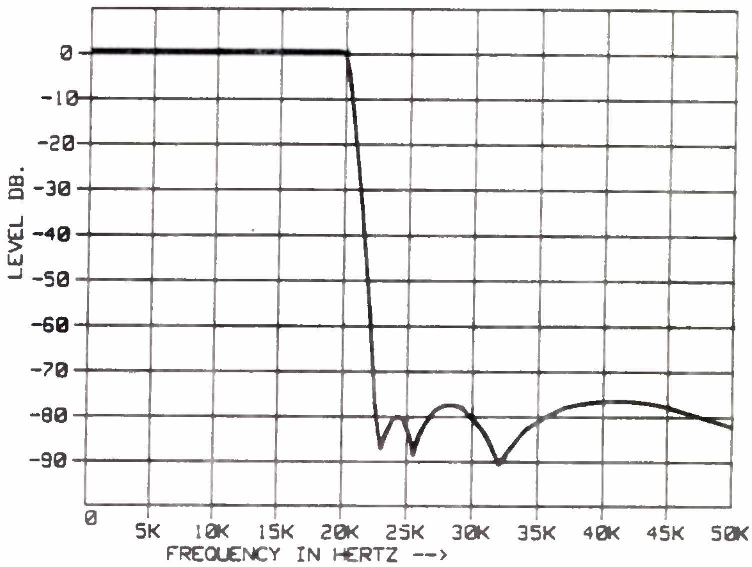


Figure 4 Typical response of an eleventh order elliptical anti-aliasing filter.

Beyond being difficult to implement, a filter of the magnitude of an eleventh order elliptical has the side effect of creating phase shifting of the signal in the area of the rolloff point which is phase distortion. The steepness of the filter is necessary because of the close proximity to the highest audio frequency we want to preserve (20 kHz), and the unwanted frequencies above 24 kHz. If the cutoff were not so steep, we could design a filter with a constant group delay which is correctable. Suppose we were able to move the sampling frequency up an octave in our analog to digital converter to 96 kHz. Then we could allow frequencies up to 48 kHz to enter the sampler without aliasing, giving our filter more room to work and greatly simplifying the filter design. This is the oversampling process. Oversampling can also be employed in the D to A converter, effectively moving the image frequencies out further, accomodating a less steep anti-image filter.

There are a great number of lengthy technical papers available on the subject of oversampling. The concept is not new, and yet there is certainly no proliferation of hardware occuring in the industry for implementation into systems. First, the application of 2X oversampling (96 kHz) offers no substantial benefits beyond the simplification of the filters. In the D to A converter, oversampling at 4X (192 kHz), along with the application of linear phase digital filtering techniques and a simpler analog filter will net us about 6 dB in signal to noise performance and reduce the phase distortion substantially. We are currently pushing the state of the art in the sampling circuitry considering the accuracy required in the A to D and D to A converters. The technology of sampling at higher frequencies is under development and may soon be realizable.

Now lets put all of these pieces together and review our digital audio system (Figure 5). In the A to D conversion we first low pass filter the incoming audio signal to be digitized with the anti-aliasing filter, removing any frequencies over half the sampling frequency. Then the signal is sampled and quantized into digital data words for storage. Playback involves converting the digital data words back to individual analog voltages in the D to A converter. These voltages are then processed by a sample and hold circuit to bridge the gap between voltage samples, and then low pass filtered by the anti-image filter to remove high frequency energy components (Figure 5).

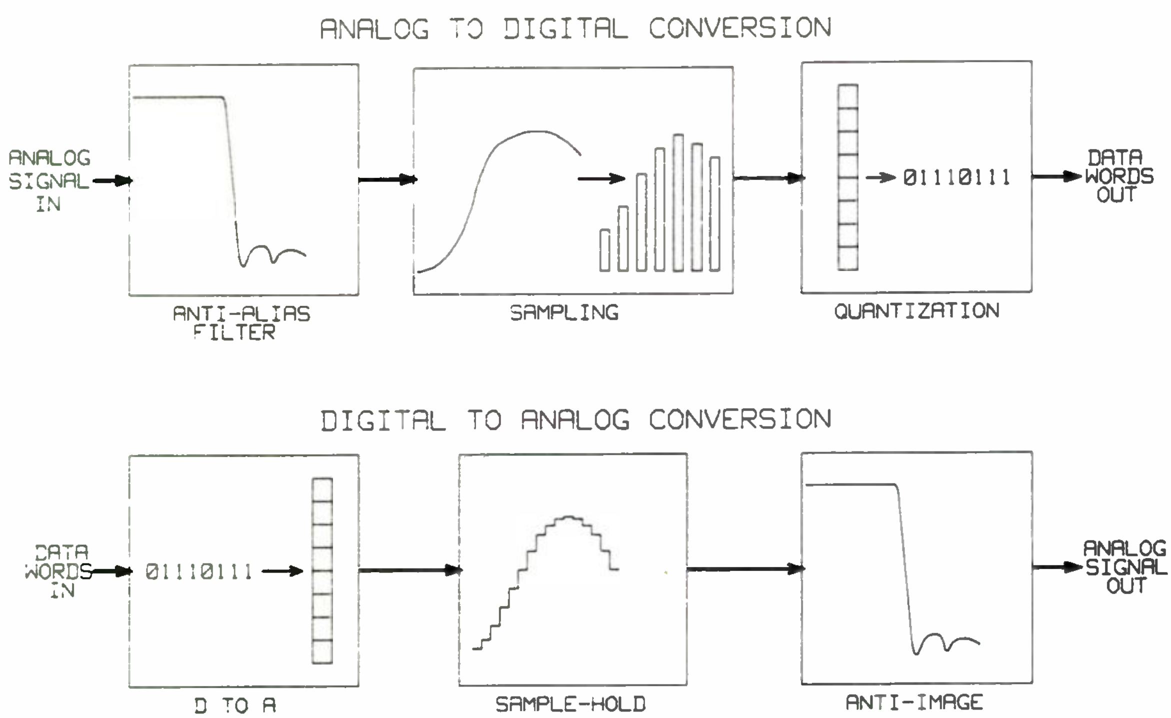


Figure 5 The building blocks of a digital audio system

STORING THE DATA

To analyze the storage requirements of a digital audio system, we will sample the audio waveform 48,000 times per second and use a 16 bit word to define the audio level in each sample. These are the basic numbers that make up professional systems. The number of bits generated in real time is 48,000 X 16 = 768,000 bits per second per channel. A stereo system runs at 1,536,000 bits per second. Data capacities are generally measured in Bytes, which is an 8 bit word. Therefore, our stereo system generates 192,000 Bytes per second of data. A minute of digital audio requires over 11 megabytes of storage. An hour is nearly 700 Megabytes. That's 575 of the high density 5 1/4 inch floppy discs. Clearly, digital audio requires an inordinate amount of storage. Where do we store all of this data? Let's look at some commercially available means. The pricing listed is for media alone, and is estimated.

MEDIA	STORAGE	TIME	APPROX COST
5 1/4 FLOPPY	360 kB	1.9 Sec	$ 1.50
5 1/4 Floppy	1.2 MB	6.3 Sec	3.50
3 1/2 Floppy	750 kB	3.9 Sec	1.50
3 1/2 Floppy	1.5 MB	7.8 Sec	3.00
Bernouli disc	20 MB	104 Sec	100.00
Optical Draw 12 in disc	2500 MB dbl side	3.75 Hr	450.00
3M HCDA 3000 Digital Audio Cart	330 MB*	22 Min	30.00
Winchester removable	20 MB	104 Sec	100.00
Winchester disc drive	320 MB	28 Min	2,000.00

* Unformatted storage capacity

Affordable data storage media will only come from a marketplace that consumes large quantities of units. The broadcast marketplace will not be capable of supporting the development of a unique media in a form factor designed strictly for their needs due to the relatively low volumes involved. In this respect, we are forced to go to the computer data industry to seek out an appropriate media to store our many bytes of data. The application of any current standard data product to digital audio comes up short of fulfilling our needs in one fashion or another. Floppy discs are short on storage and take too long to read; and a room full of Winchester drives won't fulfill a radio station's storage needs. Some manufacturers are experimenting with data reduction techniques which involve processing the digital audio data mathematically in an attempt to boil down the quantity of data to fit within the confines of existing media. To accomplish this need reductions on the order of 4 to 8 in data densities and to date these techniques haven't proven to be effective without a substantial loss in performance.

Fortunately, the computer data industry appears to be headed towards larger quantities of data stored in smaller packages and this will prove to be the shot in the arm for affordable digital audio for the broadcaster. Advanced error correction codes and media like those employed in the 3M HCDA 3000 Digital Audio Cartridge system will be the methods by which these increased capacities are accomplished.

While digital audio offers us an opportunity for vast improvements in the quality of recording and storing audio signals, it is not without its imperfections and implementation difficulties. These problems are not insurmountable, however, and digital audio will take its place in the broadcast environment. These fundamentals are intended to supply the broadcaster with an introduction to the basic concepts of digital audio to help him evaluate the various systems that are already hitting the marketplace.

TELEPHONE HYBRIDS - WHAT THE CAN AND CANNOT DO

By John Cheney

President, Comrex Corporation

I will make the assumption that, if you are attending this engineering session or taking the time to read this paper, you know what a telephone hybrid is and why your station needs one. Further I would guess that, at some time in the past, you have had occasion to use one or more and already know that they can be "sometimes" devices.

I will not bore you with a dissertation about the circuitry of telephone hybrids; there are plenty of technical articles available. I will address the problem of variability of results and will attempt to explain why telephone hybrids work well in some applications and not in others. Emphasis will be placed on the use of telephone hybrids in talk show systems, although other applications will be covered.

First I would like to talk about the telephone system itself. For all practical purposes, it is only the subscriber loop running from your station and the central office in which that line terminates that has any importance in this discussion. It consists of a SLIC (Subscriber Line Interface Card) at the C.O., your loop and a connector at the end of the loop. The SLIC serves to provide the BORSCHT functions, which are:

B - Battery feed
O - Overvoltage protection
R - Ringing
S - Supervision
C - Coding
H - Hybrid
T - Test

The subscriber loop is usually made up of a twisted pair of #24 or #26 gauge wire. The average length of a subscriber loop is 4000 feet but may be as long as 15,000 feet. If the line is greater than 6000 feet, it is often loaded to improve high frequency transmission (and to provide complications for telephone hybrids). At the station end of the loop, the telephone company will provide either a punch block or a modular connector. The battery voltage is on the order of 48 to 52.5 volts dc. Ringing voltage is usually 90 volts RMS, 20 Hz but other voltages and frequencies may be employed by different telephone systems.

The hybrid function is included in a SLIC to convert your "two wire" subscriber loop to "four wire." The send and receive ports of the hybrid are then connected to digital encoder/decoders for transmission to other central offices through other CODECs and SLICs to other subscriber loops and callers.

Once your call has been digitized, you have no further "metallic" connection to another caller's line and the balance of your hybrid will not be influenced by the electrical characteristics of other caller's lines. This was not the case with "crossbar" or "step by step" offices which provided direct connection to other callers' lines, resulting in impedance disturbances on your line with attendant upsetting of balance. It is true that some of the older systems are still in place, but they are fast disappearing.

Referring to Figure 1, you will see the average line levels which are to be expected on your loop. The signal level which you should send out is -9 dBm or about 276 millivolts. The levels which you will receive will vary between -15 dBm and -40 dBm. The noise on a line will normally be about -57 dBm. Most local calls will come in at -16 dBm but, since the telephone routing system is computer controlled, even local calls can come in at the lowest level. Usually, long distance calls arrive around -30 dBm.

Moving to the hybrid itself, Figure 2 contains circuit diagrams of several different hybrids. Operation of these various types is well covered in the literature. Regardless of the type, the performance of all hybrids is determined by the ability of the balance network to mimic the impedance of your loop at each frequency in the audio band.

TELEPHONE LINE LEVELS

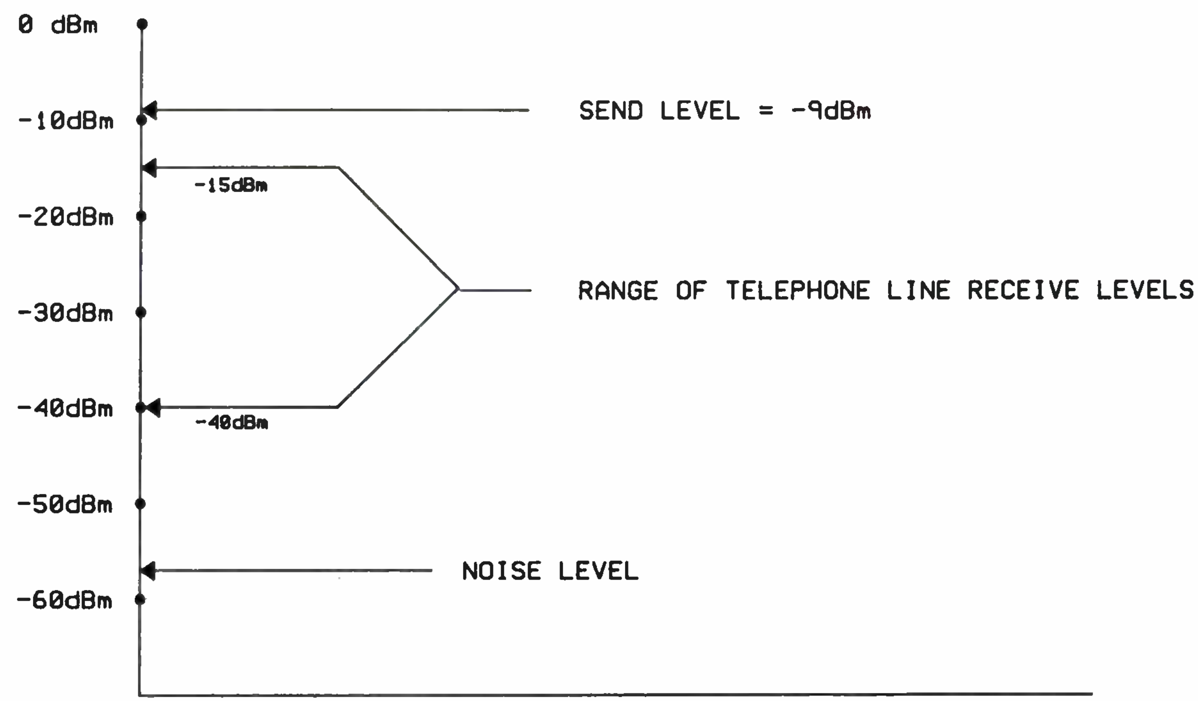


FIGURE 1

HYBRIDS

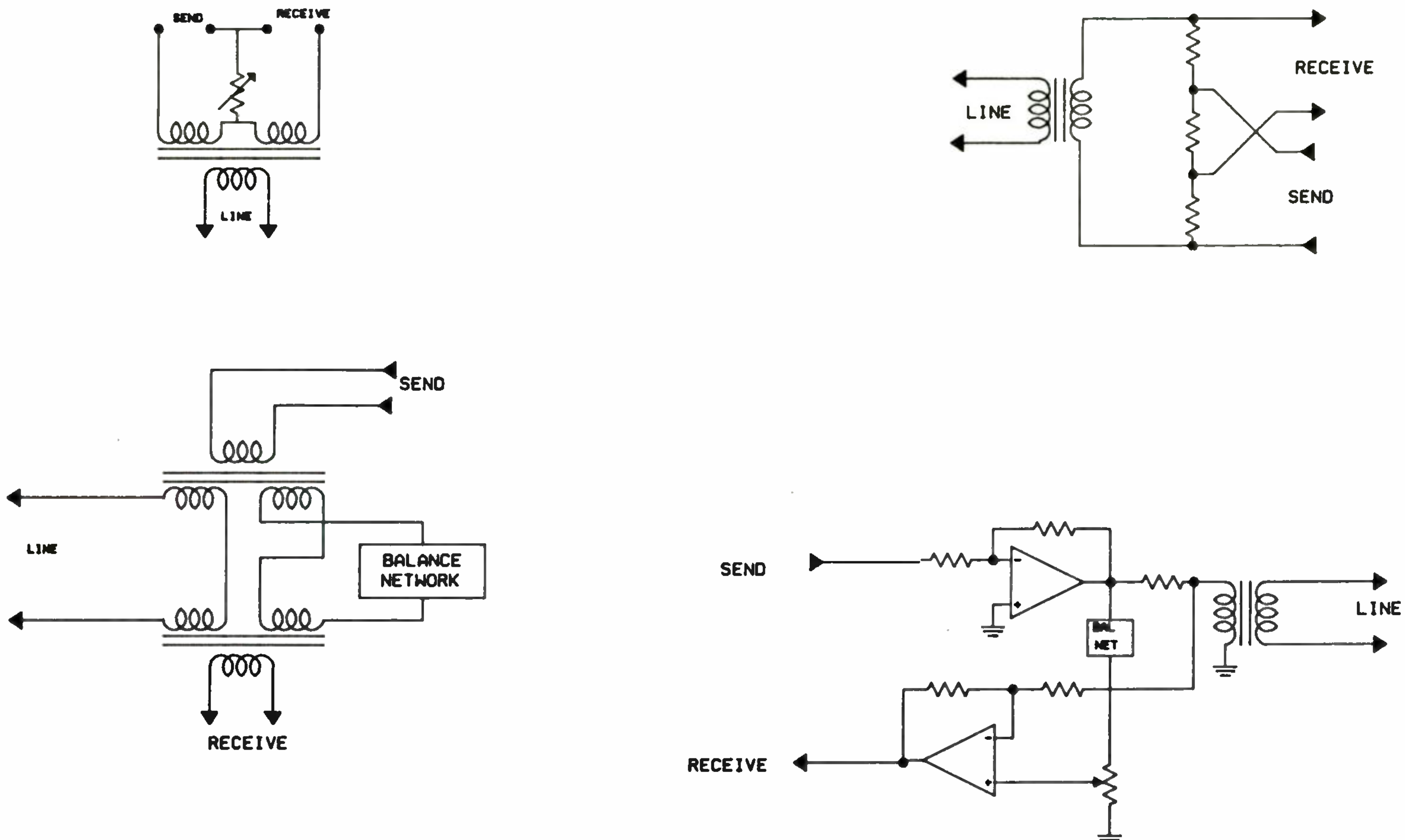


FIGURE 2

Since any line can be thought of as an infinite number of series inductances, shunt capacitances, series resistors and shunt conductances, it is easy to understand that the impedance of the loop will be different at each frequency. If you would have perfect balance at all frequencies in the audio band, you will need a network which will present exactly that same impedance at each frequency. If you require 40 dB separation between the send and receive ports, then you will need a network which will be within 1% of the line impedance, everywhere in the band. For 30 dB separation, equality to within 3% is needed and 20 dB calls for +/- 10%.

At any one frequency, the impedance of a line can be duplicated with a resistor and a capacitor and quite deep balances can be obtained. Unfortunately, at another frequency, the capacitor and resistor will have to be different. The type of null that can be obtained with a single capacitor/single resistor balance network is shown in Figure 3. The reference line at -9 dBm is the level which is sent on the line and the curve shown below is the output of the hybrid which results as the frequency is varied. As you can see, there is a 50 dB null at 650 Hz but only 11 dB at 300 Hz and 3 KHz.

Incidentally, a more popular name for hybrid null is trans-hybrid loss. As used in our company, this is the difference between the -9dBm which should be sent on the line and the level of that same signal appearing at the receive port. It may be assumed that the gain in the incoming direction to the output port is 0 dB. A hybrid is a directional coupler and, ideally, the gains from the send port to the line and the line to the receive port should be 0 dB. We then look for a high attenuation from the send port to the receive port.

Also shown in Figure 3 are trans-hybrid loss curves for a hybrid which uses only one resistor in its balance network and one which uses two resistors and one capacitor. Figure 4 shows the loss curve for a multi-pole network. It can be seen that quite respectable balances can be obtained with multi element networks.

Manufacturers of hybrid units have approached the balance problem in various ways. Several use servo systems to change resistance and capacitance values to effect maximum balance at the dominant frequency content of the program material. In

TRANS-HYBRID LOSS PERFORMANCE

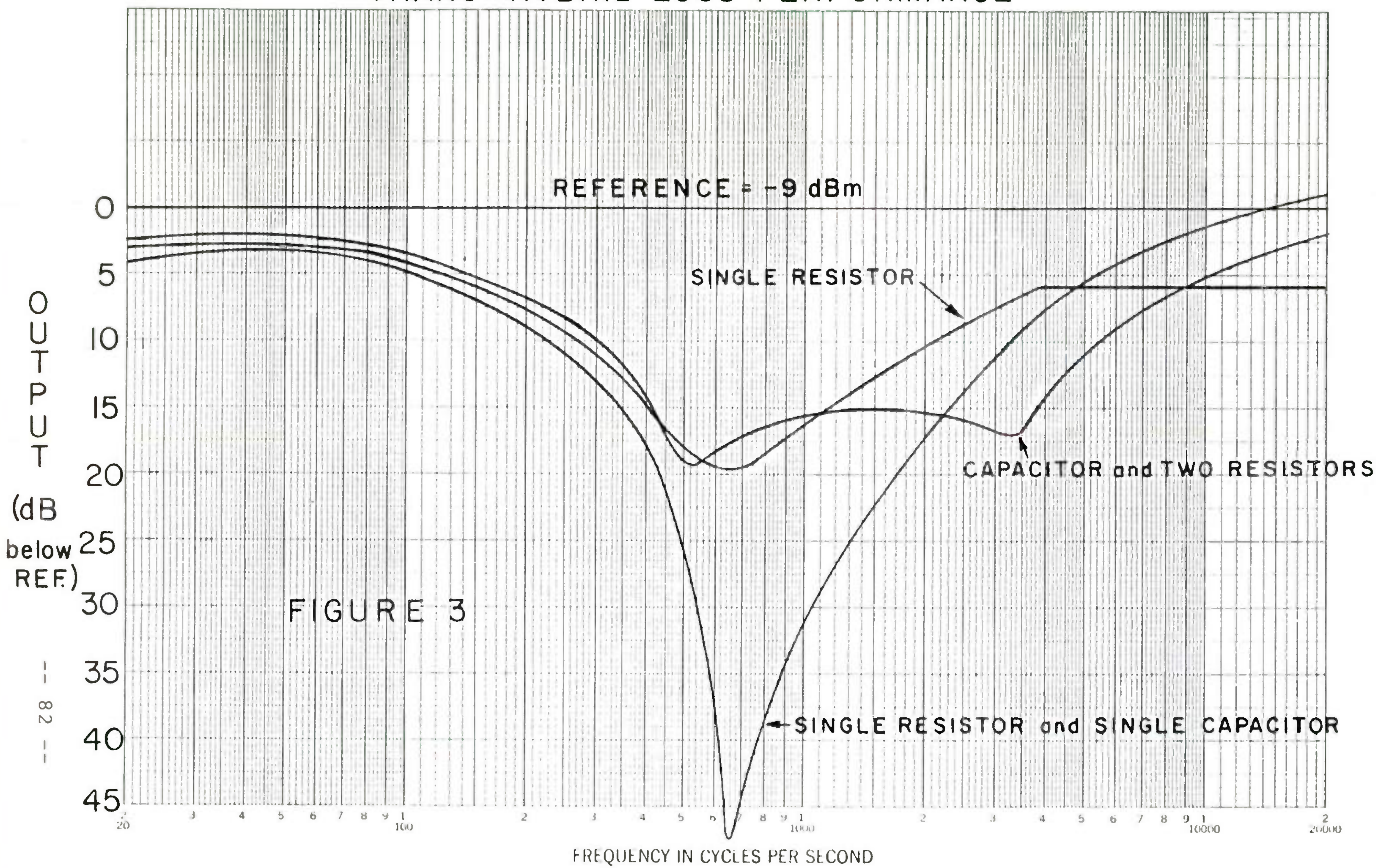
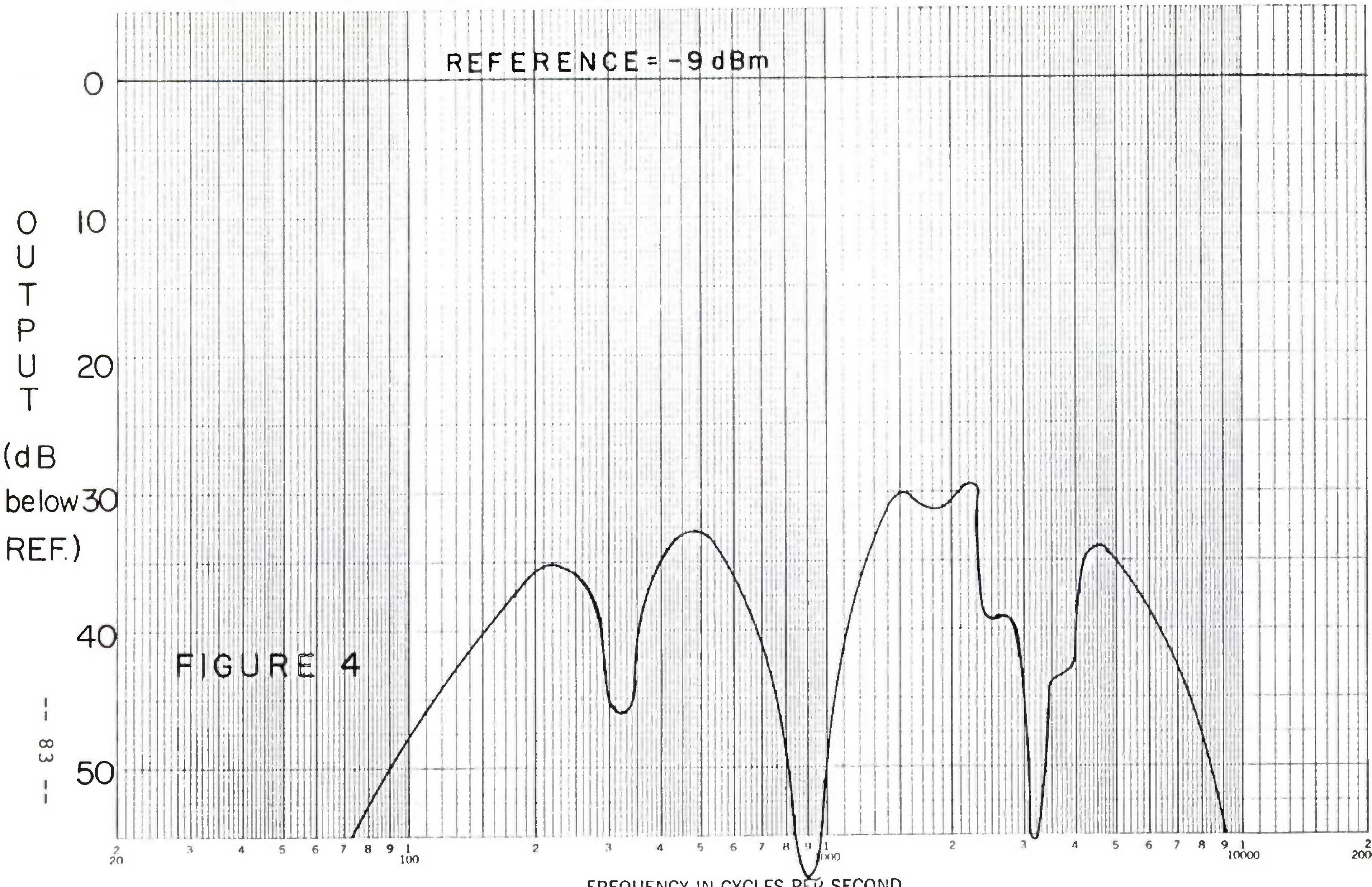


FIGURE 3

HYBRID WITH MULTI-ELEMENT BALANCE NETWORK



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effect, the notch moves up and down on frequency as required. At least two broadcast manufacturers have introduced digital hybrids which use adaptive digital filters to take the measure of the program being sent out and then to reconfigure themselves so that anything which comes in on the line that looks like what was sent out is subtracted, leaving only the incoming caller. Reports say that these work quite well. Trans-hybrid loss figures which I have heard are in the 20 to 30 dB range.

A principal broadcast need for hybrids is in talk show systems. Figure 5 is a synoptic diagram of a two line conferencing talk show system. Each line is terminated in a hybrid. The two lines may be dedicated or may derive from a key system. The mix which is put on the air is the sum of the host audio and the two callers. Each caller hears the mix minus himself. Therefore, caller 1 hears caller 2 and the host, caller 2 hears caller 1 and the host. If mix-minus circuitry is not used to separate the caller from the mix sent back to him, then oscillation and/or echo would result. The gain block shown here may be manually or automatically controlled.

Figure 6 is a system diagram of a single line of the talk show system and the host audio. Shown on the diagram are various signal levels which will be encountered. For a local caller, the incoming signal will be taken as -16 dBm and the outgoing level will be the optimum -9 dBm. At the receive port, two signals will appear. One is the incoming caller at -16 dBm and the other is the "leak" through the hybrid of the host audio. This will be -9 dBm, attenuated by the amount of the trans-hybrid loss. If we assume that -20 dB is a reasonable figure, then the undesired leak will be -29 dBm. If the call is long distance, then the incoming caller may be received as low as -40 dBm, with the "undesired" host leak through remaining at -29 dBm.

Let us examine the effects of the above. To simplify the picture, the talk show system has been redrawn and is seen in Figure 7. It is obvious that the "air" signal will be the sum of the incoming caller, the host and the host leak through. If the spectral content of the hybrid leak were equal across the band and not phase shifted, then the air signal would be the caller and the host, with the host simply amplified according to the amount of the leak. Unfortunately, what exits from the receive port is the host audio altered in frequency response and phase. In Figure 7, the effect is shown in schematic form as the host audio passed through a notch filter whose response matches the trans-hybrid loss response. We will assume that the amplitude response of the filter implies its phase response.

CONFERENCING TALK SHOW INTERFACE SYSTEM

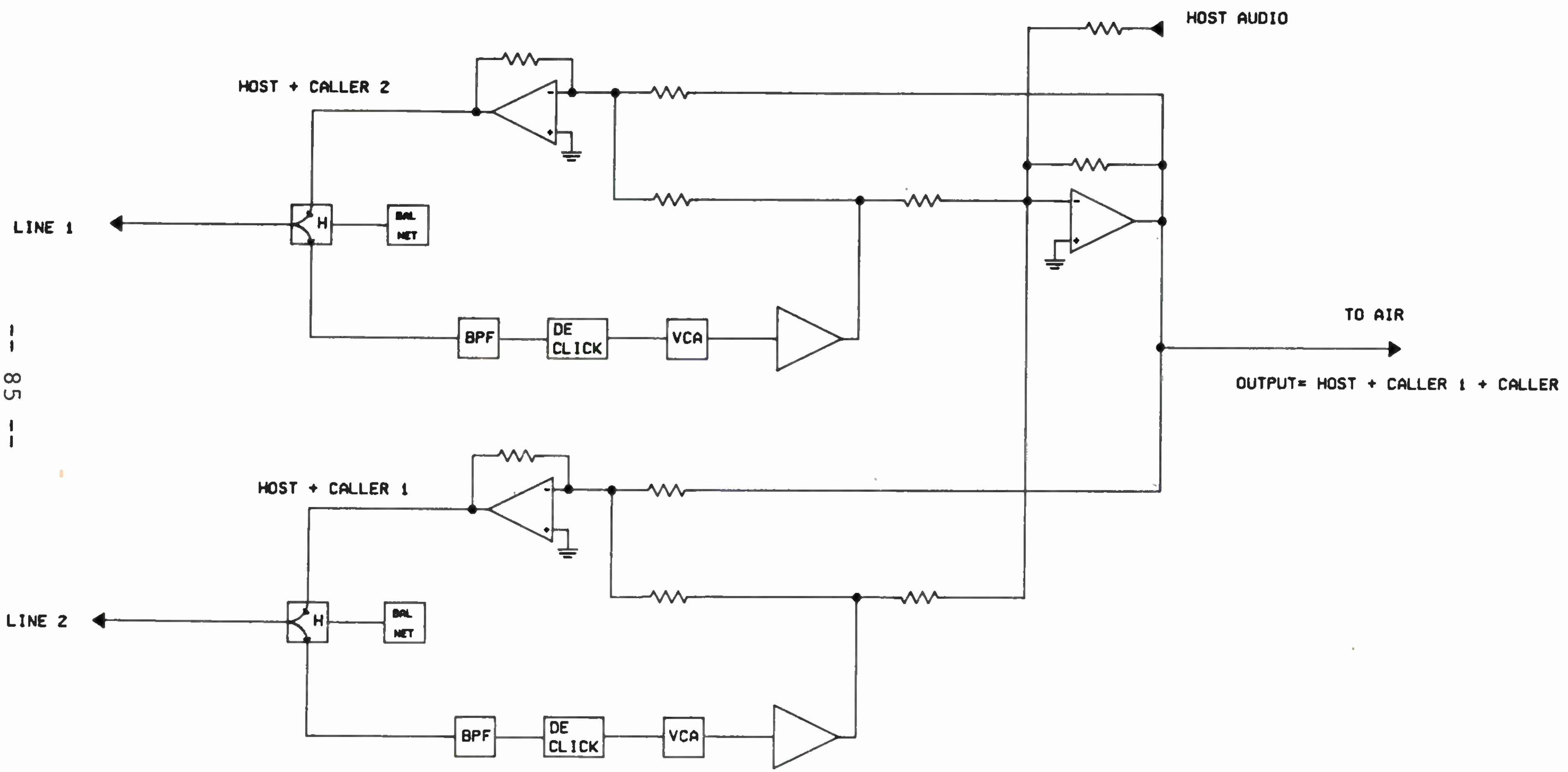
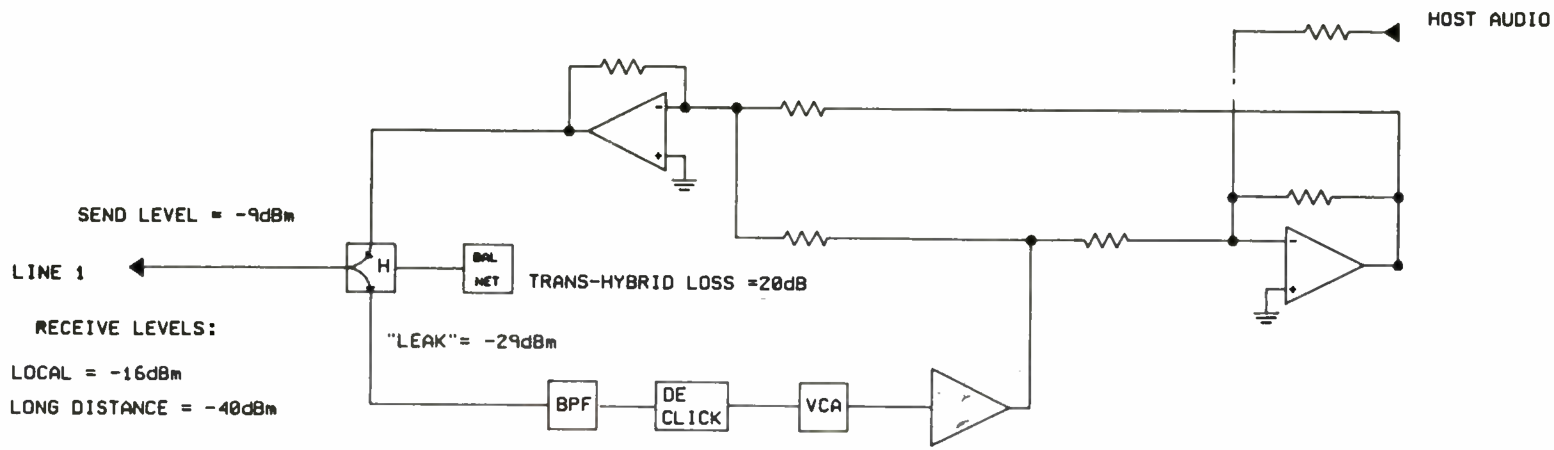


FIGURE 5

SINGLE LINE TALKSHOW SYSTEM- LEVELS



LOCAL CALL--- -16dBM (CALLER-DESIRED) + -29dBm(UNDESIRED) DESIRED/UNDESIRED= 13dB

LONG DISTANCE CALL----- -40dBm(CALLER) + -29dBm("LEAK") DESIRED/UNDESIRED= -11dB HOST SOUNDS LIKE HE OR SHE IS SPEAKING FROM "RAINBARREL"

FIGURE 6

TALK SHOW SYSTEM

HOST FREQUENCY RESPONSE DISTORTION MECHANISM

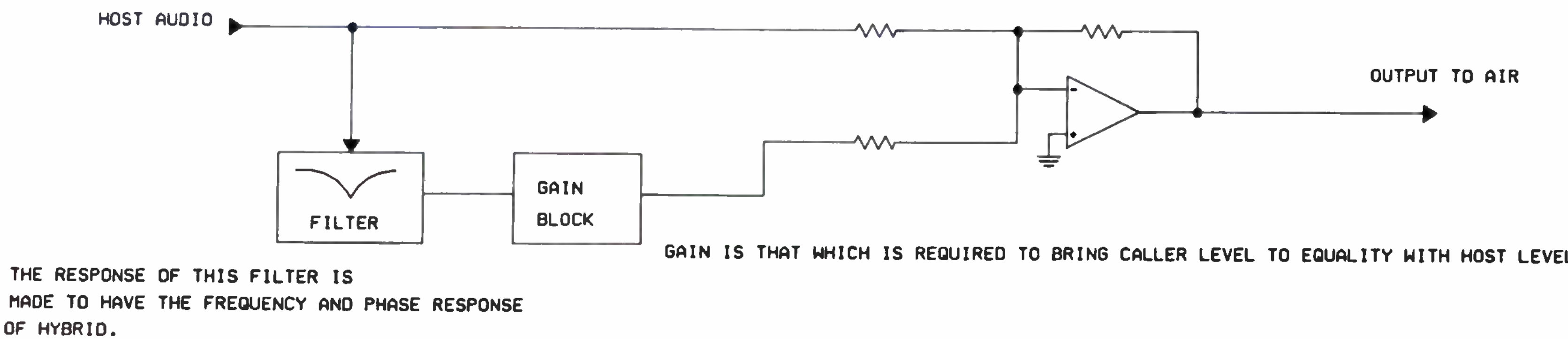


FIGURE 7

From simple arithmetic, if we sum two equal voltages, the some will be 6 dB above either voltage. If we sum two voltages whose ratio is 3.862, the sum will be only 2 dB higher in value. Another way of saying this is if an undesired signal is 12 dB lower than the desired signal, then its effect on the sum will be only +/- 2 dB.

Returning to Figure 7 let's see what happens with some real world hybrid performance figures of, say, 20 dB separation. If we assume that a local caller signal will produce -16 dBm and the host audio level to air is 0 dBm, then we will need to add 16 dB gain to make the caller and the host have the same level. The output of the gain block will then consist of a desired caller audio of 0 dBm and an undesired amplitude and phase altered host level of -13 dBm. We will assume that the caller is not speaking, so the summed output will be a host audio, altered in response by about .75 dB. The effect of the trans-hybrid leak on the host sound would be for the most part unnoticeable.

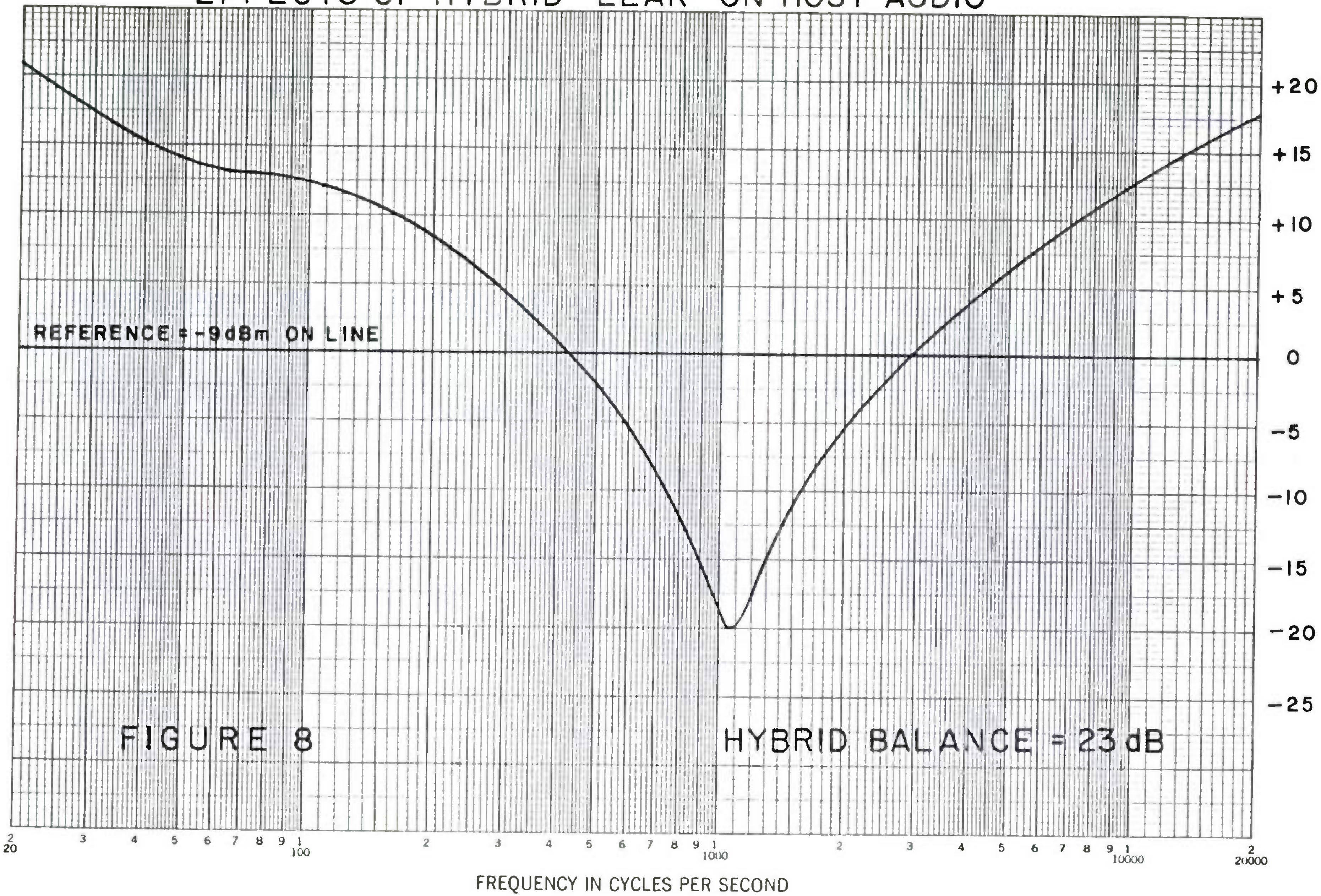
On a poor local call or average long distance call, the effect will be seen to increase. The -30 dBm call will need 30 dB gain. The -29 dB leak through of the host will be increased by the same 30 dB and will be +1 dBm at the mixing point. The host audio response will be altered by +/- 6.5 dB and this will be quite noticeable.

Now for a worst case example. The caller signal comes in at -40 dBm and 40 dB of gain is required. The undesired output at the mix point is +11 dBm. Adding the host audio and the leak through of the host audio, the resulting frequency response departs from flat by +/- 13 dB.

It can be seen from the above examples that there are very good reasons for the complaint that hybrids cause the host to sound as though he or she is in a "rain barrel." Figures 8 and 9 are plots made on local and long distance calls. You can see what happens to the host's audio.

Also, from the above, it can be seen that the actual trans-hybrid loss provided is a very important specification. There are many manufacturers of hybrids and there are just as many stated specifications. Attention should be paid to the method by which the measurement of the hybrid is made. First, the measurement should be made while connected to an actual telephone line. Artificial lines and resistors may make the hybrid look good, but your concern is what it does on your line or lines.

EFFECTS OF HYBRID "LEAK" ON HOST AUDIO



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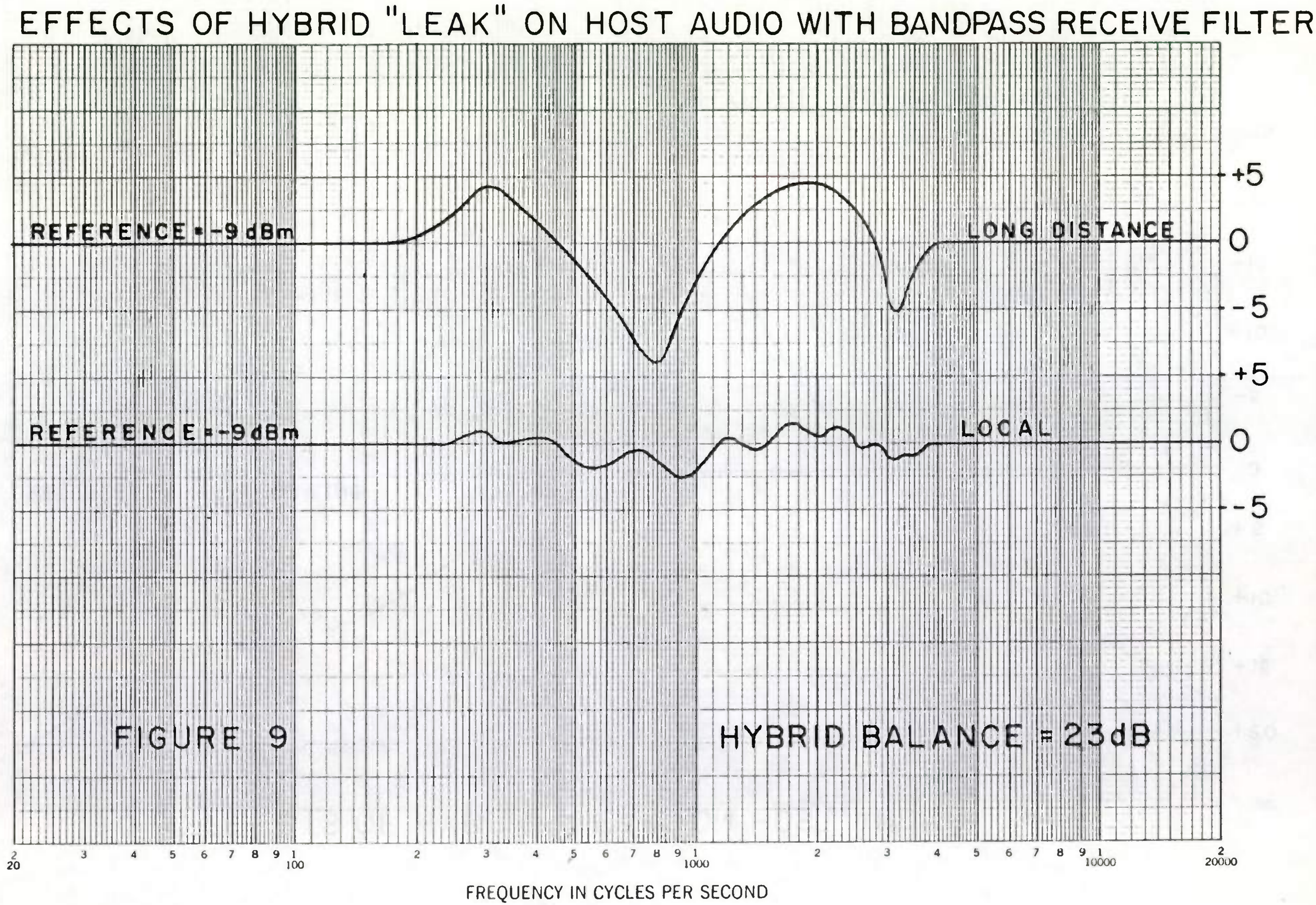

EFFECTS OF HYBRID "LEAK" ON HOST AUDIO WITH BANDPASS RECEIVE FILTER
REFERENCE = -9 dBm
LONG DISTANCE
REFERENCE = -9 dBm
LOCAL
+5
0
-5
+5
0
-5
FIGURE 9
HYBRID BALANCE = 23dB
20
100
1000
10000
20000
FREQUENCY IN CYCLES PER SECOND

--06--

The fairest way of comparing hybrids is to use a noise generator and a spectrum analyzer. If you use a sine wave sweep, the automatic hybrids can readjust themselves as the input frequencies change and give false readings. Our procedure is as follows: Sufficient input signal is applied to the send port of the hybrid to produce -9 dBm on the telephone line. Then the level at the receive port of the hybrid is measured before the gain block. This should result in a "real world" measurement of the directivity of the hybrid.

Let us assume that, in actual operation, a good hybrid will produce a fairly flat consistent null of 20 to 30 dB over a range of 300 to 3000 Hz. How do we produce a talk show system which will perform well at all incoming caller levels?

The host audio leaking through the hybrid could be solved by reducing the send level, but the caller would have problems hearing the station.

The remaining choice is to control the gain between the hybrid receive port and the mix point while the host is speaking. The obvious and most effective way of doing this is to use a push-to-talk switch. This is a reliable and inexpensive solution, but it does have the drawback of relying on the host's expertise.

To accomplish automatic control of the receive gain, a device is needed that will recognize the fact that the host is speaking. This is no easy task. A speech detector must have a threshold built in. If the audio level is above that threshold, it will operate and if it is below, it will not operate. Extraneous noise sources must be ignored. Involved decisions must be made as to timing so that the air mix is not "choppy." One can therefore appreciate the difficulty in executing a design which will work reliably under widely varying conditions. Because of the desire for automatic operation, most manufacturers have included speech detectors in their units, but you should be aware that this is not an unmixed blessing.

The following are some collateral issues which I would like to touch on.

Bridged taps

Part of basic "telephone hygiene" is making sure that your telephone lines (and especially the ones you use on air) are free from bridged taps. If you send a frequency sweep out one line and back into another line at your station, you should see a fairly flat response from 300 to 3000 Hz or better. The presence of a bridged tap will produce a marked roll off in the high frequency response.

The telephone companies construct their outside plant (cables, poles, junctions boxes, etc.) so that when a customer asks for telephone service it can be provided quickly without having to run in new cables or lines. They keep track on how given areas are growing and attempt to accurately anticipate customer needs. Whenever they install a cable, they provide excess pairs.

One the of techniques used is to install cables following major arteries and their cross streets. Whenever these cables cross, a junction box is provided. If new service is requested, the telephone company looks up the site on their cable map and makes the connection at the appropriate junction box. Later, if that service is stopped, the telephone set is removed and service on that line is denied by a digital command to the SLIC at the central office. If another new service request comes for a site a couple blocks down, the telephone company may use the same line on the main artery and simply connect to the junction box two blocks down. If they don't remember to disconnect the first cross street cable, the new line will consist of a cable between the central office and the new site PLUS an open ended piece of cable going down the first cross street.

This is called a Bridged Tap and it is quite possible to have more than one on a line. Because it is open ended and not terminated, it is simply a large capacitor connected across your line. It's a great source of noise and its capacitance is more sensitive to the environment then a discrete capacitor would be. Also, its presence will make hybrid balance difficult.

Although the telephone companies are not forced to remove these taps, we know of many stations who have been successful in their art of gentle persuasion. We suggest that you expend whatever effort is needed to be sure that the lines you use on air are free of bridged taps.

Automatic vs fixed hybrids

If you intend to use one hybrid and connect any one of several lines to it, then you probably will want to use an automatic hybrid. Probably one of the digital hybrids would be best for your use. These units work well - and they are also relatively expensive.

The approach our company recommends is the use of a dedicated hybrid card for each line which may be in your station. When a line is not being used for talk shows, contests, news, etc., it rests connected to your in house telephone system and used for the regular conduct of business. If a call comes in or is made for air or recording, a digital logic command transfers the call from the in house telephone system to the hybrid which has been specifically balanced for that line. The hybrid then presents that line in four wire form to the studio where it may be connected into the mix-minus matrix and placed on air. One advantage of this system is that each line may be routed throughout the station and used as four wire in multiple locations (talk show studio, news room, production studio, etc.).

Conferencing of lines

If you intend to conference callers with the host and each other, you have to decide whether you will be satisfied with the performance obtained when you simply parallel two lines or whether you require a more sophisticated approach. If you parallel the lines, you will have difficulty setting gains. (Picture a -16 dB call paralleled with a -40 dB call.) Also, when you parallel two or more lines, your hybrid will have to rebalance for the new line impedance it will see. Even the best automatic hybrid will be hard pressed to adjust for a line which changes impedance that much.

The full conferencing approach provides a separate hybrid for each caller with all callers and the host being brought to equal level and appropriate mix minus applied before their audio is combined.

In conclusion, the subject of telephone hybrids and telephone interface is far too involved to treat comprehensively in one short paper. I hope, however, that I have given you some insight into what telephone hybrids can and cannot do. Whatever need you have for a telephone hybrid, may I suggest that you take the time to analyze your system requirements before making a purchase. It won't be wasted time.

FIBER OPTICS: THE MEDIUM OF CHOICE

by C. Robert Paulson

Director of Product Management, Artel Communications Corp.
Hudson, MA 01749

Fiber Optics? The Medium of Choice?

"Fiber Optics?? That's point-to-point stuff! It can't be switched. That's no good for broadcasters."

"Fiber Optics?? It's great stuff! But it's not available from here to where I want to send my signals."

"Fiber Optics?? It has a lot of characteristics that are better than copper! But it's too expensive."

Or, "It's too difficult to pull." Or, "It's not fully developed yet." Or, "________." (Fill in your own reasons for viewing fiber optics as just another new technology, maybe useful someday.)

"But it's not a right-now, results-justifiable, cost-justifiable solution to any of my in-plant routing or inter-plant network distribution television signal transmission needs!"

Is that what you're saying, Bunkie?

"Balderdash!" says I.

Copper Versus Microwave: A Features and Capabilities Brief

Copper is common. It's an old friend of broadcasters. But it does have shortcomings as a television video and audio signal transmission medium. Otherwise you wouldn't be using microwave radio, both terrestrial and satellite routes for shipping and receiving your signals in lots of diverse applications: STLs, Common Carrier feeds, ENG pickups, Satellite News Relays Networking ----.

"Microwave gets my signals where they need to go, <u>without</u> pulling cables or paying telco charges. It's the answer to copper's deficiencies!"

"Is that what you're saying, Bunkie?"

"Balderdash!" says I.

Microwave circuit use by broadcasters has therefore "burgeoned" in recent years. The broadcast microwave "private property " bands from 2 GHz on up and over 40 GHz are now bursting with broadcasts, narrowcasts, multicasts and point-to-point casts.

Ah----but there's the rub! The <u>burgeoning</u> use and <u>bursting</u> channel loads make you want to <u>bludgeon</u> the people who are spreading QRM all over your signals, always at the most inappropriate times.

True? Ergo, now that you're relying heavily on microwave to do what copper can't do, or does poorly, you're beginning to look for ways around microwave's problems.

Copper and/or Microwave Versus Fiber:

A Features and Capabilities Comparison

Please note: Fiber optics isn't <u>any</u> of those words and phrases I put in your mouths in the opening section. Your video, audio and data signals, modulated on light traveling in fiber, are---

(1) SWITCHABLE!

Not as easily as switching electron currents in copper, true. But the technology for switching in light is figuratively moving toward product almost as fast as the speed of light.

And while you're waiting for optical switching to arrive, you can switch signals transported by fiber by converting them to their original electron current basebands, and then switching them in the equipment you already own. Your received signals will still be better quality than if you had transported them via copper.

Further, if you use technology-matched routing systems from Dynair, and Tx/Rx terminals from Artel, you can begin immediately to benefit from the advantages of shipping your signals on optical light carriers traveling in glass fibers.

The benefits?

MORE bandwidth, link capacity, SNR, multiplexability, circuit reconfigurability, etc.; and

ZERO signal quality deterioration from EMI and RFI, or radiation which creates interference or makes easy the non-intrusive tapping of supposedly secure transmission channels.

(2) FIBER IS OUT THERE.

According to industry execs writing in the August 1987 Communications News "Special Report on Fiber Optics Technology," AT & T and at least 15 other "long distance" service providers have buried over 28,000 route miles of fiber in U.S. soil in the last few years. It criss-crosses the 48 contiguous states in seven major trunk routes, with hundreds of spurs in all directions.

The number of miles of individual fibers on these routes is already between the 10^8 and 10^9 orders of magnitude. In the next few years, a 20 percent increase to 35,000 route miles is predicted.

Each of the seven trunk routes finds AT & T fighting with two or three or four or more competitors, including their former friends the RBOCs. (The homophonic similarity of the erstwhile "Baby Bells" acronym to running shoes in this competition with their former Ma is curiously refreshing! It's nice to see the old matriarch trying to stay ahead in a marketing race!)

Growth of the long-haul fiber plant has slowed appreciably since mid 1985. This has unnerved some manufacturers of fiber, who evidently built manufacturing capacity in expectation of a 100 percent takeover of long-haul copper by fiber before 1990!

That won't happen. But on the other hand, don't assume that fiber will never replace copper 100 percent as the exclusive medium for either long-haul or short-haul video, audio and data information transmission.

Fiber is now taking over from copper in many short-haul applications important to broadcasters. More on that later. But first, know and believe that ----

(3) FIBER IS COST-COMPETITIVE.

Cost and pricing trends in lasers, LEDs, detectors, and all the electronic components that are built into fiber optic system transmitting and receiving terminals, are now on the annual cost shrinkage curve well-known in the computer industry.

"Yes, but ---," you interject. "Most of those components are also used in microwave and copper system terminals, so they're also falling in price."

"Yes, but ---," I reply. "Let's look at what fiber terminal interconnection does for you that copper and microwave interconnection can't match."

First, fiber's bandwidth (circuit capacity) characteristics are similar to if not better than microwave's. Certainly fiber is better than microwave in the relatively miniscule transmitter power needed to push the same amount of information the same distance in earth surface miles, as compared to satellite circuits. And the fiber circuits are EMI and RFI free, with no foreseeable limit on the number of circuits which can be set up.

And fiber's information carrying capacity is already thousands of times greater than copper's, given the same cable diameters.

Further, there is no foreseeable limit on the amount of information one fiber can carry, looking at the capacity increases now being offered via FDM, TDM and WDM techniques.

Frequency Division Multiplexing -- modulating each signal onto its own discrete carrier frequency, then combining these carriers into one signal which modulates the light beam;

Time Division Multiplexing -- Sequentially sampling a parallel array of digitized signals at a rate many times higher than the maximum bit rate of any one of them, and modulating the light beam with that bit stream.

Wave Division Multiplexing -- Merging two or more independently modulated light beams of different light wave lengths (hue or frequency) onto one fiber, all propatated in the same direction, or alternatively some propagating in the opposite direction to provide Full Duplex (FDX) links.

Truly, broadcasters, fiber IS your medium of choice for both in-plant routing and inter-plant distribution of your television (composite or component analog or digital VIDEO, multiple channel analog or digital AUDIO and ancillary audio or digital DATA) signals. Fiber IS "right now." It IS "results justifiable" and, "cost justifiable."

Building Fiber Bridges Over "Broadcaster's Chasm"

For most of this decade, broadcasters have been operating astride a widening chasm caused by a multi-epicentered earthquake. On one side of the chasm, technological adventurists are tirelessly creating new analog and digital signal forms for transporting video, audio and data from chassis to chassis in the same equipment rack, from equipment to equipment in the same room, from room to room in the same building, from building to building, from city to city, etc.

Considering all the alternative video, audio and data signal forms and quantities of channels alternatives, the number of possible combinations of signals which can require simultaneous, co-routed, time-synchronous, phase-synchronous transportation over one circuit can number in the tens of thousands!

Discouraging! No!

Worse -- the problem is simply impossible to cope with, with any currently available or even dreamed-about copper or microwave radio multiplexed signal transmission systems.

And that's the other side of the broadcaster's chasm: the deficiencies in quantity, quality, reliability, utility, versatility, etc., of those copper and microwave systems, in the face of burgeoning signal form diversity and channel quantities growth curves.

In-plant copper transmission systems aren't adaptable to changing needs caused by signal form proliferation. Inter-plant copper/microwave systems have many shortcomings depending on their location and application -- cost, availability, vulnerability, inflexibility, capacity----.

What's the only transmission medium that can bridge this broadcaster's chasm?

Fiber!

Fiber pathways, interconnecting terminals in which incoming video, audio and data signals are conditioned and then multiplexed, using Frequency Division, Wave Division and Time Division multiplexing techniques, before transmission over a single fiber.

That's right -- a single fiber, over distances as great as 50 to 60 kilo-meters (30 miles or more) before repeatering is required. Repeatering (light amplification and retransmission) without need to decode the signal group to baseband!

Further, that single fiber, even as it's passed through repeaters, can carry bi-directional signals by using laser light of different frequencies as the signal carrier in each direction, as explained previously.

Fiber System Availability

While broadcasters have been spread-eagled astride their chasm, more and more desperately searching for techniques to close it up, the erstwhile copper and microwave common carriers and their fiber and terminal suppliers have actually been creating them.

That's right. Those accursed common carriers once known collectively as "Ma Bell" are posing and posturing and pitching their willingness to become broadcasters' partners again! Since "Divestiture Day," they've become a growing, seething, crawling, intertwined welter of Baby Bells and RBOCs competing with each other across old territory boundaries. Plus non-regulated "OCCs" (Other Common Carriers), Teleports, Interconnects, Bypass Services, Satellite Networks, et al.

Some if not yet all the companies in this competitive melange are ready and willing to quote you on transmission services organized exactly per your specification. Like "dark fibers," for instance, to which you connect your own terminals. And they're ready, willing and able to provide those services on your timetable!

Unbelievable? Not at all. Give them a call!

Multiple carriers competing on diverse routes! Ready to do your bidding!

What more do you want to hear before you "make a decision for fiber" for your long-haul transmission?

The "AN" Alphabet: LBCMRNI!

Most broadcasters have an essentially uniform answer to the previous question: "Our network, and lots of special services networks available to us, take care of our long-haul signal importation and delivery needs. What we're interested in is solutions to our short-haul problems -- around our coverage area, for instance, where we now have to rely on ENG microwave or tape shuttling to rush a fast-breaking story to air."

"-- And we also need solutions for our medium haul needs, where we're now using terrestrial microwave and satellite microwave to bring back our signals."

But these needs are now totally satisfiable, if you "make a decision for fiber." Satisfiable by both those manufacturers of fiber and terminals who are serving the long-haul common carriers, and also by the specialized medium-haul and short-haul transmission systems operators to whom you can turn for your ENG, EFP and SNG transmission services needs.

Whatever your haulage distance or route needs, there's an "AN" (Area Network) in the following hierarchical nesting of increasing-length transmission networks ready to be custom-created for you:

- LAN -- Local, within a room or floor in a building;
- BAN -- Building, from top to bottom floor, via a routing switcher;
- CAN -- Campus, between buildings, using either fiber pathways, or Laser/Infrared light aerial paths;
- MAN -- Metropolitan, from building to building, or campus to campus, using municipal rights of way for fiber, or free air for laser line of sight, transmission;
- RAN -- Regional, from city to city, suburbs to city;
- NAN -- National, via one of those 16 (or more) common carriers;
- IAN -- International, via TAT and TransPac submarine cables of fibers.

Copper certainly can't begin to offer you that hierarchy of expandable, interconnectable, switchable, multi-channel, signal form-transparent, upgradable hierarchy of networks. Neither can microwave, via either wave guides or terrestrial or satellite links.

Don't you think it's time at least to begin to consider "making a decision for fiber?"

A Primer of Broadcast Station Uses for Fiber Systems

Considering only short-haul systems (LAN, BAN, CAN, and MAN Networks), for transporting NTSC video/MTS Audio from place to place, the list of fiber systems alternatives to copper and microwave is certainly attention-getting! STLs; ENG pickup and delivery; fibering up stadiums, arenas, legislatures, city halls, concert halls, etc. with "FiberTrax®," the triax alternative; microwave downlink and uplink interconnect; in-plant long hauls from routing switcher crosspoints to locations too far away for coax to handle.

Obviously, it's time for broadcasters to make a decision for fiber for short-haul and medium-haul, "broadcast standard" signal transmission, as well as committing to it for long-haul services.

A Primer of Production/Post Production House Uses for Fiber Systems

Facilities involved in production and editing of television commercials and programs have now become experimental laboratories for some of the myriad of other-than-broadcast-standard signal form alternatives. These real-world signal transportation needs include component analog or digital video (including HDTV), four channels of 90 dB SNR digital audio, plus assorted channels of ancillary data. Existing routing and distribuion systems designed to handle an NTSC video and one single analog audio channel are useless!

What's the only transmission medium ready to handle any conceivable assortment of those transmission needs?

Fiber!

It's also time for your decision, you new wave producers and editors of spots and programs.

Making the Decision

Contact the manufacturers of fiber and terminals. Contact their VARs (Value Added Resellers -- more familiar to broadcasters by their traditional descriptions as "Distributors," "Dealers" and "Systems Houses"). Contact several of that welter of Ma Bells and their burgeoning, brawling, unregulated competitors.

You'll be absolutely amazed at how easy it's become to "make a decision for fiber!"

UHF MULTI-CHANNEL TV ANTENNA SYSTEMS

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Introduction

Regulatory pressure is increasing to limit available transmitting site locations. The FAA restricts the locations for tall towers. The EPA/FCC RF radiation guidelines increases the difficulty of locating new transmitting systems. An economical solution is the use of a single, multi-channel antenna and a single transmission line to maximize the number of stations at a given site.

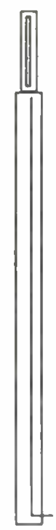
This paper discusses the advantages of using a single antenna and transmission line for high power, UHF transmission systems. The types of antennas suitable for broadband, multi-channel application are presented. Antenna performance capabilities in terms of patterns, VSWR, and power handling are reviewed. The bandwidth limitations and design requirements of waveguide transmission line, components and transmitter combiners are covered.

Advantages of Single Antenna and Line

The advantages of common sites for multiple channels are easily recongnized. Real estate and tower costs are lower. Using a location already accepted by the FCC, FAA and local governments provides faster and smoother approvals. All broadcasters benefit from the common orientation of receiving antennas.

For most broadcasters, the major consideration is economic. The use of common antenna and transmission line hardware further reduces the costs in comparison with multiple-antenna installations. The multi-channel transmission system can even reduce the cost of the tower relative to mounting separate transmission systems on a common tower. Figure 1 compares a dual-antenna installation and a multi-channel antenna system on one tower. Even with the additional cost of the transmitter combiner, the savings using the broadband, multi-channel system are significant. Although this example compares the case of two channels, the more substantial savings available by adding additional channels to the multi-channel system is obvious.

There are some applications where a broadband antenna system is the only solution. Existing structures are often limited in available aperture and/or windload capacity for antenna mounting which prevents the use of multiple-antenna installations.



BUDGET COSTS

Dual Antenna		Dual Channel
610	Tower, 1200'	500
400	Antennas	250
300	Waveguide, WR1500	150
-	Combiners	35
1,310	Totals (k $)	935

Savings $ 375 k

FIG. 1 Cost comparison of dual-antenna vs. dual-channnel.

Antenna Considerations

Broadband vs. Narrowband

Multiple channels on a single antenna require the antenna to be broadband in both pattern and impedance (VSWR) characteristics. As a result, the antenna design represents a significant departure from the narrowband, single channel pole antennas that are commonly used for UHF. The typical single channel UHF antenna uses a series feed to the individual radiating elements while a broadband antenna has a branch feed arrangement. The two feed arrangements are shown schematically in Figure 2.

At the design frequency the series feed provides co-phased currents to its radiating elements. As the frequency varies the electrical length of the series line feed changes such that the radiating elements are no longer in-phase outside of the design channel. This electrical length change causes significant beam tilt out of band and an input VSWR that varies rapidly with frequency.

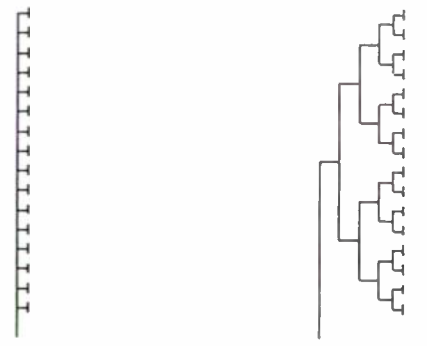


FIG. 2 Series feed vs. branch feed.

By contrast the branch feed configuration employs feed lines that are nominally of equal length. Therefore the phase relationships of the radiating elements are maintained over large frequency spans. This provides vertical patterns with stable beam tilts as required for multi-channel applications. Broadband VSWR performance is also possible since the input impedance is essentially the average of all the radiating element impedances.

The basic building block of the multi-channel antenna is the broadband panel radiator. The individual radiating elements within a panel are fed by a branch feeder system that provides the panel with a single input cable connection. These panels are then stacked vertically and arranged around a supporting spine or existing tower to produce the desired vertical and horizontal radiation patterns.

This design provides great flexibility for accommodating the multiple requirements of gain, power handling, and horizontal radiation pattern shape. For increased gain, stack more panels vertically. If greater power handling is required, use more panels either vertically and/or around the mounting structure. Since there is freedom to vary the phasing, power division, and pointing direction of the panels, numerous custom horizontal radiation patterns are possible.

It is this flexibility that makes it difficult to put absolute limits on the performance capabilities of multi-channel antenna systems. The scope of this discussion is limited to the higher power systems which require waveguide for the power transmission from transmitter to antenna.

The ability to combine multiple channels in a single transmission system depends upon the bandwidth capabilities of the antenna and waveguide. The antenna must have the necessary bandwidth in both patterns and impedance (VSWR). It is possible to design UHF antenna systems for lower power applications using coaxial transmission lines that provide whole band capability. For the high power systems, the waveguide bandwidths set the limits of channel separation.

Pattern performance is not the limiting factor. As frequency increases the horizontal pattern circularity deteriorates, but this effect is generally acceptable. Also, the electrical aperture increases with frequency which decreases the vertical pattern beamwidth. If a high gain antenna was used over a wide bandwidth, the increase in electrical aperture might make the vertical pattern beamwidth unacceptably narrow. This is not a problem with the channel limits set by the waveguide.

For the antenna, it is primarily VSWR that limits its bandwidth. Broadband VSWR performance is accomplished by the following measures:

a) branch feed system
b) broadband individual panel VSWR
c) elements fed with phase pertubation scheme[1]
d) properly located discontinuities in feed system[1]

Item c), phase pertubation, generally occurs as a natural byproduct of providing null fill and beam tilt phasing over an antenna's vertical aperture. As a result the similar impedance characteristics of the panels cancel out at the antenna's input. It is also possible to incorporate phase perturbation within individual bays of an antenna to improve the impedance characteristics of the bay. By properly locating discontinuities in the feed system, item d), the impedances of the multiplexed channels are matched and an improvement in overall bandwidth occurs.

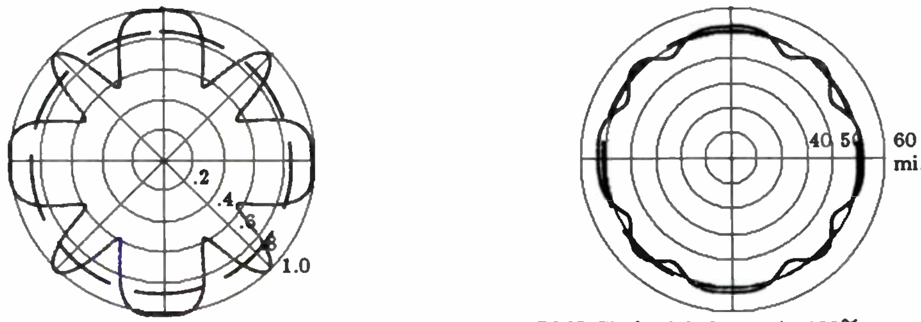
By use of the above techniques, it is possible to offer an antenna input VSWR specification for each of the combined channels that assures excellent picture performance comparable to that achieved by single channel antennas. A typical VSWR specification is:

Antenna Input VSWR :	1.05:1	Visual carrier
(Each channel)	1.10:1	Visual upper sideband

Horizontal Pattern Capabilities

Due to the physical design of broadband panel antennas their cross-section is larger than the typical narrowband pole antenna. Therefore, as the operating frequencies approach the high end of the band, the circularity (average circle to min or max ratio) of the omnidirectional antennas generally deteriorates. For example, the economical 4 panels per bay antenna configuration has typical circularities of $\pm$ 2 dB at channel 26 which increases to $\pm$ 3 dB at channel 69.

It is illuminating to consider what meaningful effect this has on coverage. Figure 3 shows a 4 around horizontal pattern with $\pm$ 2.5 dB circularity and its RMS circle. Since an omnidirectional antenna's ERP is based on the RMS gain value, the Grade B coverage distance is actually greater in the maximum directions as well as less in the minimum directions. For this case, a variation of +2, -4.5 miles relative to the FCC calculated Grade B distance of 51.5 miles (1200 ft. hgt.; 2.5 MW ERP) is obtained. The total coverage area calculates to within 1.1 % of the RMS circle coverage area on a square mile basis. In essence the area of service is nearly the same as a perfectly omnidirectional antenna. If an area of higher population exists, the pattern maxima could be orientated in that direction.

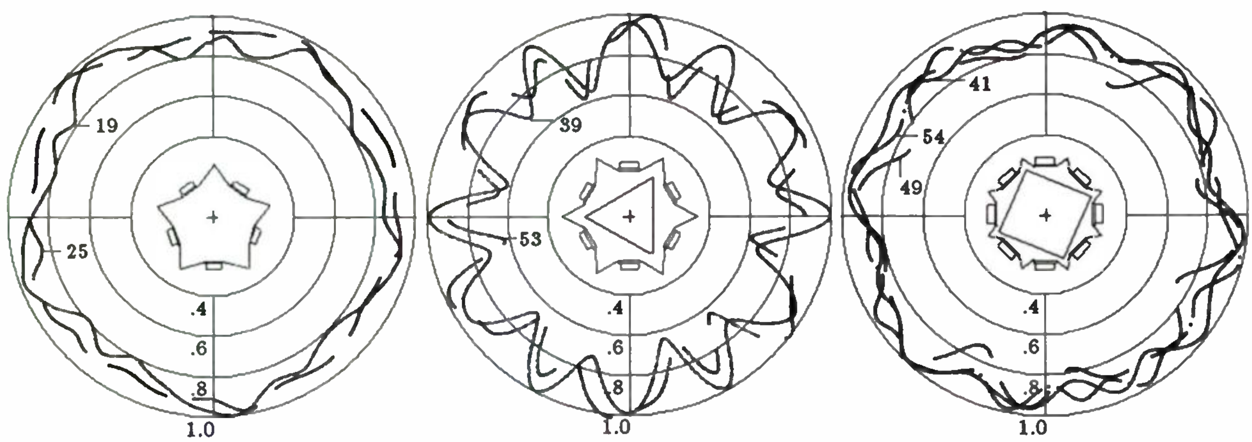


+1.3 dB, -3.7 dB
relative to RMS circle

RMS Circle: 8,348 sq. mi., 100%
Hor. Pattern: 8,257 sq. mi., 98.9%

FIG. 3 Horizontal pattern vs. coverage.

Improved circularities are possible by arranging additional panels around the supporting structure. Previous installations in Canada and England have used 5, 6, and 8 panels per bay. These are illustrated in Figure 4 along with measured patterns at different operating channels. These approaches are often required for power handling considerations especially when three or four channels are involved.



CH: 19 & 25
$\pm$ 1 dB

CH : 39, 43, 47, & 53
$\pm$ 2.1 dB

CH : 41, 45, 49 & 54
$\pm$ 1.5 dB

FIG. 4 Measured patterns; 5, 6, & 8 panels per bay.

The flexibility of the panel antenna allows directional patterns of unlimited variety. Some of the more common applications are shown in Figure 5. The peanut and cardiod types are often constructed on square support spines as indicated. Another type of cardiod is possible by side-mounting on existing triangular towers.

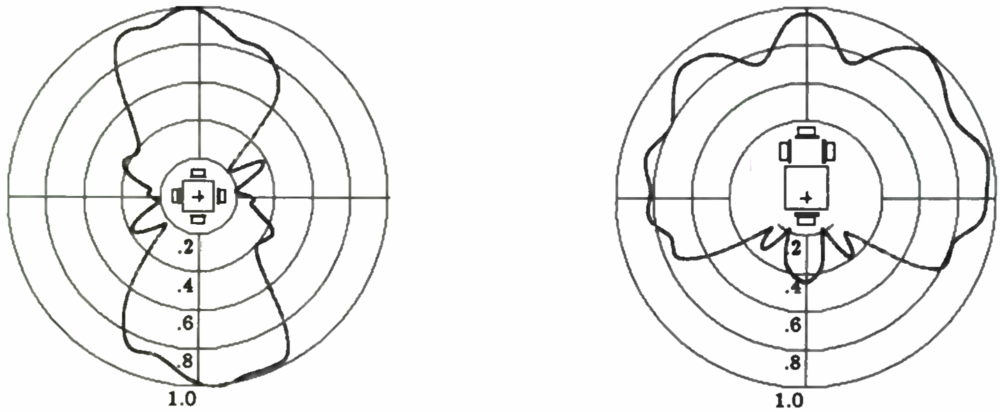


FIG. 5 Measured Peanut and Cardiod patterns.

Also indicated in Figure 6, is the capability to provide different horizontal radiation patterns for each channel. This is done by changing the power and/or phase to some of the panels in the antenna with frequency.

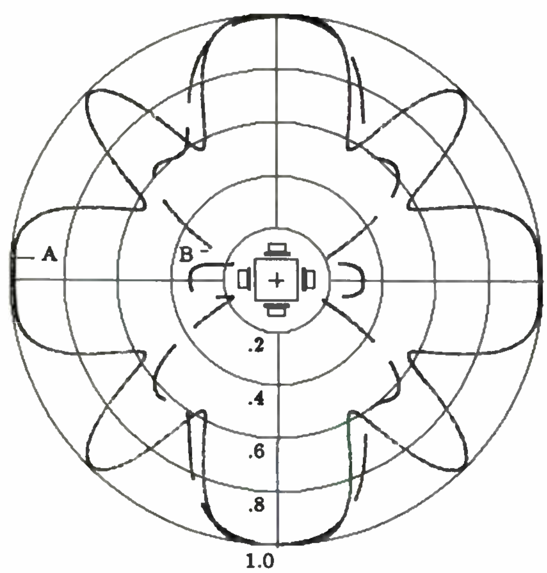


FIG. 6 Calculated patterns: CH "A" - omni
CH "B" - peanut

Most of the above antenna configurations are possible using a circularly polarized panel. This panel can also be adjusted for elliptical polarization with the vertical polarization receiving less than 50% of the power. Using a circularly polarized panel will reduce horizontally polarized ERP's by half.

A summary of the above mentioned antenna configurations is given in Table I. The pertinent performance characteristics for multi-channel applications are indicated. These include circularity for omnidirectional cases, gain, input power, and total available ERP. This is the ERP available for division among the channels operating on the antenna if maximum input power and gain were applied. If circular or elliptical polarization are involved, the ERP allotted to one channel represents the sum of the horizontally and vertically polarized ERP's.

Table I is a partial list of the possible antenna configurations. It serves as a systems planning guide only. Desired channel combinations outside the recommended spacings are possible with non-standard designs.

TABLE I

UHF MULTI-CHANNEL ANTENNA CONFIGURATIONS

OMNIDIRECTIONAL

ANTENNA/WG	RECOMMENDED CHANNELS	TYPICAL CIRC (± dB)	MAXIMUM RMS GAIN (PWR)	MAXIMUM POWER (kW)	AVAILABLE ERP (kW)
4 PANELS per BAY					
WR-1800	14-39	2.0	31	230	7130
WR-1500	30-56	2.5	31	210	6510
WR-1150	52-69	3.0	31	195	6045
5 PANELS per BAY					
WR-1800	14-39	1.0	31	285	8835
WR-1500	30-56	1.5	31	265	8215
WR-1150	52-69	2.0	31	245	7595
6 PANELS per BAY					
WR-1800	14-39	1.5	31	345	10695
WR-1500	30-56	2.0	31	315	9765
WR-1150	52-69	2.5	31	290	8990
8 PANELS per BAY					
WR-1800	14-39	1.0	31	460	14260
WR-1500	30-56	1.5	31	420	13020
WR-1150	52-69	2.0	31	390	12090

DIRECTIONAL

ANTENNA/WG	RECOMMENDED CHANNELS	TYPICAL HRP GAIN (PWR)	MAXIMUM PEAK GAIN (PWR)	MAXIMUM POWER (kW)	AVAILABLE ERP (kW)
PEANUT					
4 PANELS per BAY					
WR-1800	14-39	2.4	74	125	9250
WR-1500	30-56	2.4	74	115	8510
WR-1150	52-69	2.4	74	110	8140
CARDIOID					
3 PANELS per BAY					
WR-1800	14-39	1.8	56	170	9520
WR-1500	30-56	1.8	56	160	8960
WR-1150	52-69	1.8	56	145	8120
CARDIOID					
4 PANELS per BAY					
WR-1800	14-39	1.8	56	230	12880
WR-1500	30-56	1.8	56	210	11760
WR-1150	52-69	1.8	56	195	10920

NOTE:

1) Max Gain - highest gain typically offered to maintain stable performance.
2) Available ERP - total ERP for division among channels if maximum input power and gain applied.
3) CP or Elliptical polarization - allotted ERP is sum of horizontal and vertical polarizations.

Transmission Line and Component Considerations

The choice of the proper size and type of transmission line and components must occur in the first stages of any multi-channel system consideration. A properly chosen transmission line system must fulfill the requirements of: propogating all channels at the same time, operating at the total combined output transmitter power, offering minimal insertion loss and VSWR on all channels, and providing high isolation between the transmitters. Guidelines which summarize these requirements have been developed to aid in the selection of a proper system.

Frequency Selection, VSWR, Insertion Loss

The first things to consider when defining a system are the frequencies to be combined and the VSWR and insertion loss specifications desired. There are two basic limitations on the allowable frequencies of a multi-channel operation; the minimum allowable channel separation needed for proper (-35dB min.) transmitter isolation by the combiner and secondly the largest channel separation at which the desired maximum VSWR and insertion loss for each channel can be obtained in the transmission line and antenna.

The first restriction is relatively constant across the UHF band for particular types of combiners. For the combiner design considered here, a spacing of approximately four channels is required between the combined channels to provide this minimum isolation. Different combining techniques might allow lower spacing if required for a particular situation, however, the VSWR and insertion loss may not be as low as that obtained in this example. The second, and more severe limitation on the frequencies to be combined in a multi-channel system is the maximum VSWR and insertion loss allowable for each channel. The VSWR performance is highly dependent on the desired system layout, number of elbows, transitions and other components needed to carry the signal to the antenna. The maximum bandwidth and minimum residual VSWR associated with different components varies widely with their design and configuration in the system. Thus all components should be considered before a final maximum system bandwidth is determined. The restriction then is based on the frequencies allowed to propagate in the waveguide and associated components of the system.

Figure 7 shows the channels over which the three standard rectangular waveguide sizes can propogate effectively and also the channels which are usually specified for single channel operation. These waveguide sizes are standard, custom sizes are available for systems whose frequencies might fall between a standard size. From this figure the range of allowable channel combinations for each waveguide size can be seen. In this figure the total range does not consider any other component bandwidth restrictions and assumes a straight waveguide with no elbows or transitions.

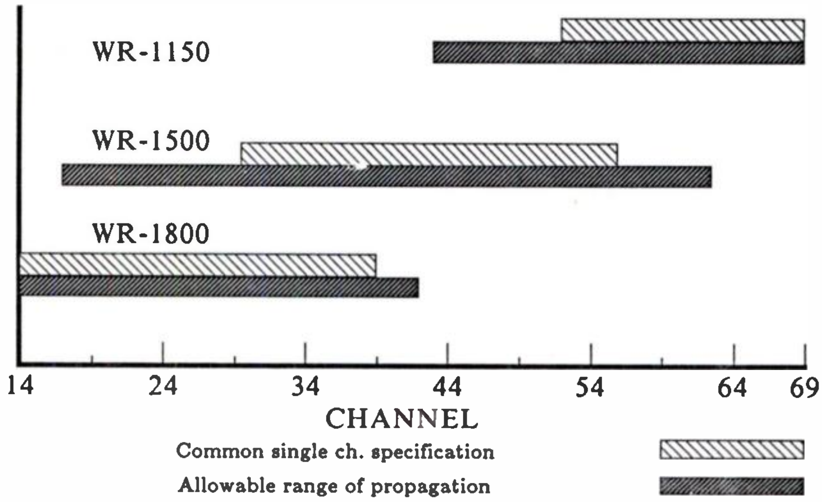


FIG. 7 Standard waveguide channel usage.

The question then is what performance can we expect for elbows, transitions and other components. A complete analysis of the bandwidth of each component in a prospective system is far beyond the scope of this presentation. Below we will generalize the performance of several important components. For our application the standard definition of bandwidth does not apply. Instead, bandwidth is considered the band over which satisfactory separate fixed tuning of each channel is possible and the channels in between are ignored. Obviously this type of tuning becomes difficult when large residual VSWR's must be tuned out on each channel. Standard waveguide transitions have been made which cover more than an entire waveguide band but have fairly large residual VSWR's (1.3:1) or do not handle large amounts of power. Optimized transitions have been designed to cover most of the waveguide band (30% bandwidth) with very good performance (<1.08:1 VSWR) at high power levels. Elbows present the other main restriction to VSWR performance. Standard single mitered elbows have untuned bandwidths (<1.08:1) of approximately 5-10% while optimized double mitered elbows can operate over a 30% bandwidth with similar performance. In summation, the total VSWR of a system can be determined after choosing the frequencies which can propogate in the waveguide and examining the number of elbows, transitions and other components which increase the systems VSWR and lower the maximum bandwidth available. If a system is designed with only one transition and one elbow a wider frequency range, nearing that of the waveguide, is available. If there are several transitions, elbows and bends then the range will be much more restricted.

Figure 8 gives the attenuation or insertion loss of these same standard waveguides over their frequencies of propogation. Although insertion loss should always be as low as possible, the restrictions placed on the choice of the waveguide size above limit the optimization of a system for insertion loss. If it is found that all channels can propogate in two different size waveguides then the size which offers lower insertion loss should be chosen. This would occur when the combined channels are spaced close together and are near the end of one size operating range and in the middle range of another.

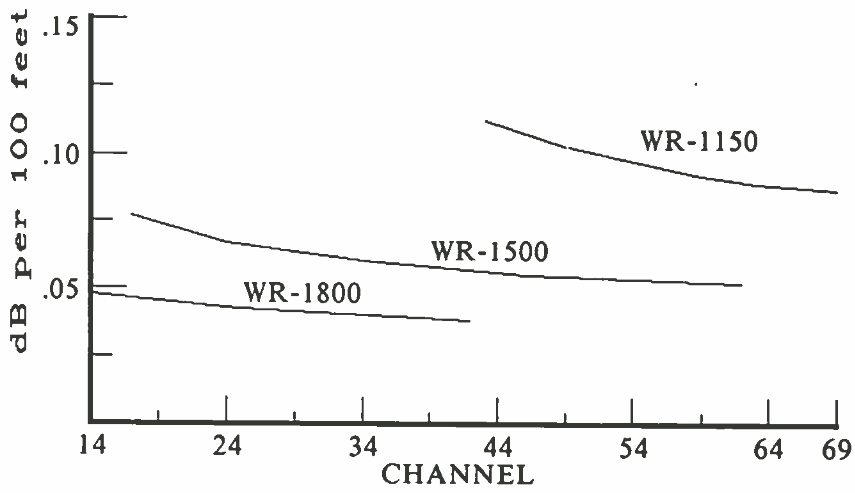


FIG. 8 Standard waveguide attenuation curves.

Total System Output Power

The system must be able to handle the combined total power of both channels. Current waveguide and components have theoretical power handling capabilities of many megawatts although only tested up to 240 kW, the current available maximum transmitter power for a single UHF station. Higher combined power levels may be possible dependent on the necessary components and frequencies.

Waveguide Section Lengths

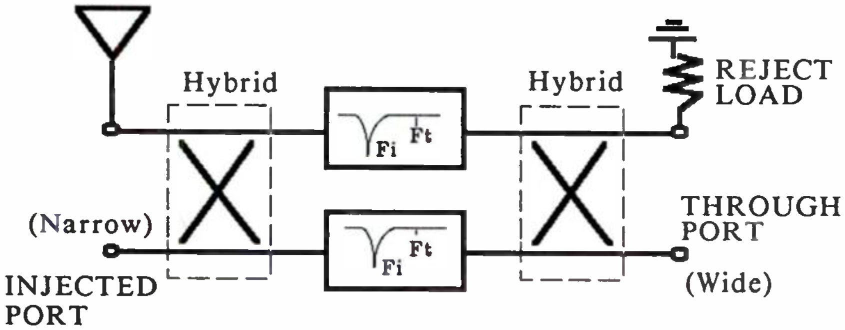
The lengths of waveguide sections in a long waveguide run must be calculated such that the reflections due to flange mismatches do not sum in phase causing high VSWR within any of the desired channels. This standard practice for waveguide installations becomes more complicated when more than one channel must be considered and may result in lengths which are not standard. A formula has been derived to aid in the proper selection of this length.

Combiner Considerations

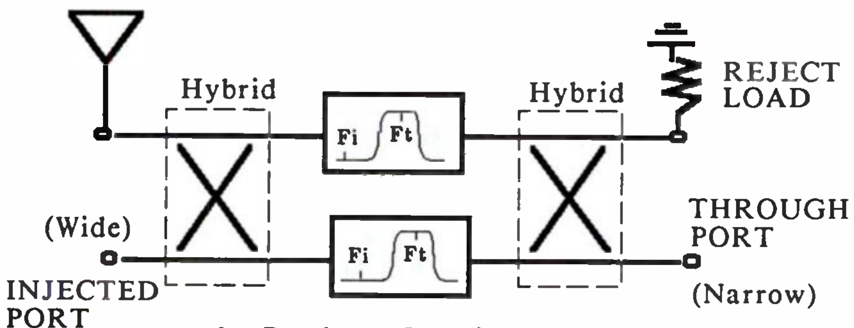
A combiner must be designed for each installation which provides high isolation, low VSWR, and low insertion loss for each of the transmitters in the system. Two basic types of combiners are currently in use for multi-channel operation. The branch type, where a separate filter is used for each input and combining occurs at a tee or common point, has the disadvantage of offering relatively low isolation between inputs and is much more sensitive to impedance changes of the source and load. The constant impedance type uses two hybrids and band-pass or band-reject filters tuned to one of the input frequencies. They offer high isolation, that of the hybrid plus that of the filter, excellent impedance characteristics, and relatively low loss.

Constant Impedance Combiners

Figure 9 shows the schematics for two types of constant impedance combiners. The choice of filtering in the combiner is dictated by the bandwidth needed (6 MHz for TV), the spacing of the input channels and the insertion loss desired.



a. Band-reject Combiner



b. Bandpass Combiner

FIG. 9 CONSTANT IMPEDANCE COMBINERS

In the band-reject combiner (CIBR), figure 9a, signals fed into the injected port are reflected by the apparent short circuit of the filter at this narrow or notch frequency and appear at the output port. Signals fed into the through port are split and allowed to pass unimpeded through the filters and also appear at the output port. This type, used in standard CIN visual/aural TV diplexers, performs well when the frequencies to be combined are closely spaced and of narrow bandwidth. To obtain high isolation over the full TV bandwidth a band-reject filter requires several cavity sections and an associated high insertion loss results for the through frequency.

In the band-pass combiner (CIBP), figure 9b, again signals fed into the injected port are reflected by the apparent short circuit of the filter and appear at the output port. Signals fed into the through port are split and now pass through the narrow passband of the filter and again appear at the output port. Thus the narrow ports of the two types of combiners are reversed, the injected port of the band-reject and the through port of the band-pass are narrow. Band-pass filtering requires somewhat wider channel spacing but offers lower insertion loss and higher isolation along with better thermal stability than that of the band-reject design. Thus the benefits of band-pass combiners make their use attractive for this application.

Combiner Performance Obtainable

The performance of the CIBP combiner can be predicted once the frequencies and the minimum isolation have been established. An isolation value of -35dB minimum is satisfactory to avoid problems of intermodulation. As stated previously four channels of space are required to ensure this isolation with the current design. A VSWR of less than 1.1:1 is obtainable for the CIBP combiner on both channels, the through port having a somewhat higher VSWR than the injected port because of travel through the filter. Insertion loss for the through port will again be somewhat higher than the injected port because of travel through the filter. This difference is minimized by design optimization of the band-pass filters for the frequencies involved. Insertion loss of less than 0.25dB for the through channel and 0.1dB for the injected channel are obtainable.

Conclusion

Multi-channel installations have been common at FM and VHF TV for many years now with excellent results. Multi-channel operation has now been introduced to high power UHF TV with similar results. We have shown the benefits of multi-channel operation, and discussed antenna, transmission line, and component considerations for a proper multi-channel system design.

Multi-channel system operation using waveguide is possible with current designs for stations which fall within approximately a 30% frequency bandwidth and are spaced more than 4 channels apart.

Acknowledgment

The authors wish to thank M. B. Anders and his staff at the Alan Dick & Company Limited for the information pertaining to broadband panel antenna performance.

References

1. M. B. Anders, "A Case for the Use of Multi-Channel Broadband Antenna Systems", NAB, 1985.

"TRUE APL PICTURE POWER OF A TV TRANSMITTER "

By Joseph J. Giardina, Thomas J. Vaughn and John Neuhaus

A new technique has been developed to measure the true APL Picture Power during normal programming of the visual transmitter.

The visual power output meter on the TV transmitter reads peak-of-sync power and not average power. The average power measured during power meter calibration does not represent the average power transmitted with normal programming.

There are a number of situations where the true APL Picture Power should be known: (a) Most meters used to measure non-ionizing radiation read average power. Prediction of power density requires the knowledge of true average ERP. Conventional average power meters cannot be used because the average power changes with video content; (b) Most RF components, including transmission line, are average power limited. When components fail it is often necessary to know if the limits of the component were exceeded.

This paper describes the theory and the measurements made at WWHT(TV) and the long term relationship found between average power and typical programming content.

The results showed that the APL Picture Power to be 4.32 dB below the licensed peak power during normal programming.

Discussion

The APL Picture Power is the average level of RF power transmitted over time . It consists of fixed parameters, such as horizontal and vertical sync and blanking and a variable parameter which results from normal video program modulation (Figure 1).

Conventional transmitter power metering systems measure peak power and cannot be used for average power measurements, since most of the energy transmitted is a variable dependent on picture content. A new method utilizing a random sampling technique was developed which permitted us to measure the power transmitted during normal programming. This system is easily calibrated against the FCC method of measuring average power while the transmitter is modulated with sync and blanking.

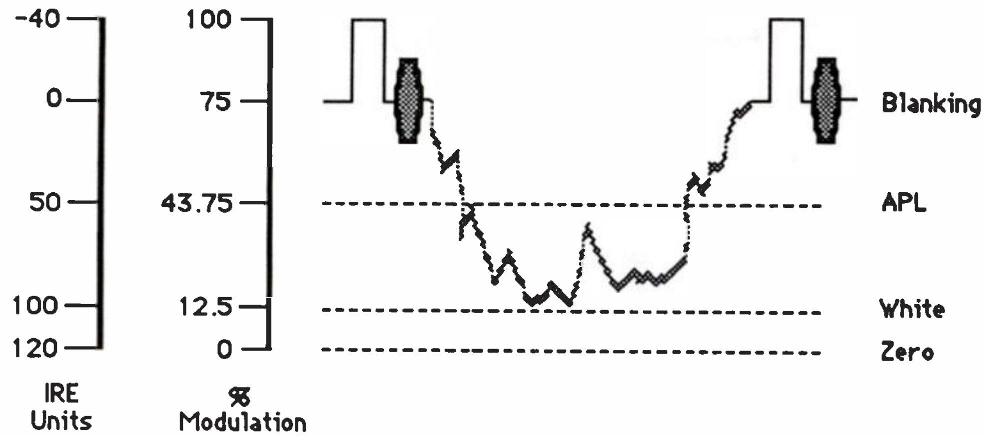


FIGURE 1

The relationship between the various terms used is tabulated below:

TABLE 1

	IRE Units	% Modulation
Peak Power	- 40	100
Blanking	0	75
Reference White	100	12.5
Zero Carrier	120	0

Method of Measurement

The output power meter of a television station's visual transmitter must, by current FCC Rules & Regulations, be calibrated with a calorimeter or against a calibrated wattmeter (1). These same rules dictate the use of a modulating signal consisting of sync and blanking during the calibration.

(A) Average Power with Blanking

The idealized horizontal portion of the modulating waveform for power calibration is shown below:

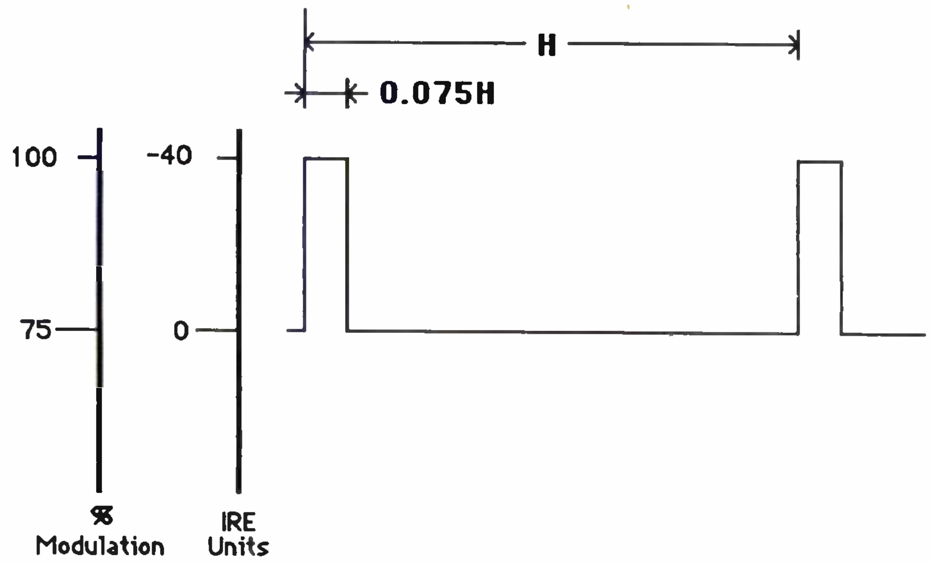


FIGURE 2

The average power is determined by measuring the RF energy converted to heat. The following formula is used to calculate average power using the calorimetric test setup shown in Figure 3 .

Average Power (kW) = (Temp (out) - Temp (in)) * Flow (GPM) * .264

The constant (.264) is determined by properties of the coolant, in this case water.

The Peak power (or licensed power) is determined by:

Peak Power (kW) = Average Power (kW) * 1.68

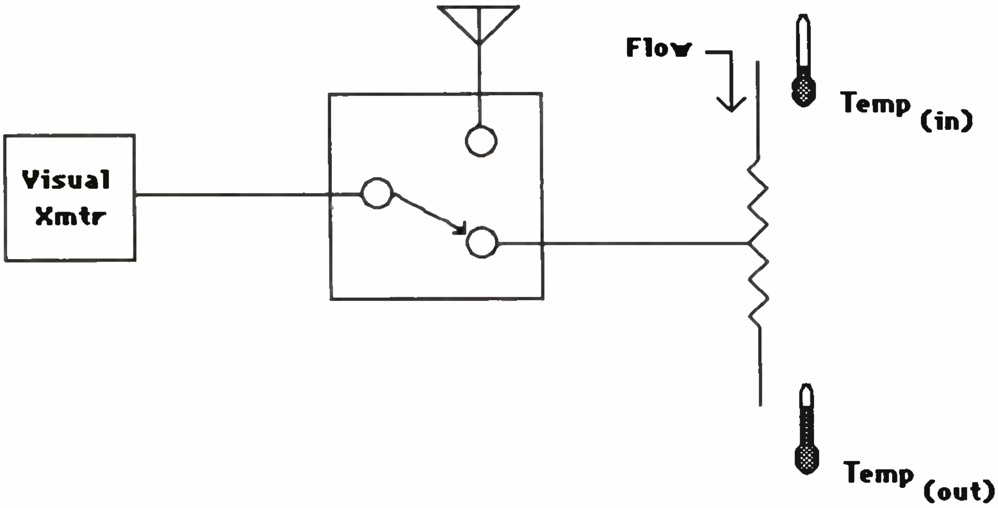


FIGURE 3

This calibration procedure is used because it is convenient, readily reproducible, and provides a precise method of power meter calibration. However, it does not accurately represent the true average power that is applied to the transmission line system and radiated by the antenna. The actual average output power of the visual transmitter is less than that value obtained during power calibration.

(B) Average Power with Program

The typical program material applied to the visual transmitter during the broadcast day consists of a video signal that varies between 7.5 IRE units (70.3125% modulation) and 100 IRE units (12.5% modulation). Average power is calculated by multiplying the percent of modulation squared by the duration of that modulation.

When the modulation consists of a flat field (Figure 2), the average power can be calculated using the equation shown in footnote (2) of the appendix. This equation reduces to:

$$A = [.1510 + .0346\,(d^2) + .7621\,(b^2)]\,C^2$$

and

$$\frac{\text{PEAK POWER}}{\text{AVERAGE POWER}} = \frac{C^2}{A}$$

Where:

A = Average power
C = 1, (Peak carrier amplitude normalized to 100%)
.1510 = Contribution to average power of horizontal and vertical sync and blanking.
.7621 (b^2) = portion of the field representing the active portion of lines 21 through 262 1/2 times the modulation squared.
.0346 (d^2) = portion of the field representing the active portion of VITS lines 10 through 20 times the modulation squared.

Table 2 lists the PEAK POWER/AVERAGE POWER values that can be derived from the above formula. Also, the decrease in average power and the corresponding relationship to licensed peak power versus the increase in modulation is clearly indicated. We can see that the average power transmitted during normal operation (3) ranges from 91.8% (1.68/1.83) to 30.7% (1.68/5.48) of the value that was obtained during power calibration.

TABLE 2

IRE UNITS POWER	PERCENT of MODULATION	PEAK POWER / AVERAGE POWER	PERCENT of AVERAGE POWER	PERCENT of PEAK
0	75	1.67	100	60
7.5	70.3125	1.83	91.8	54.6
10	68.75	1.88	89.4	53.2
20	62.5	2.14	78.9	46.7
30	56.25	2.43	69.1	41.2
40	50	2.77	60.6	36.1
50	43.75	3.16	53.2	31.6
53.75	41.41	3.32	50.6	30.1
60	37.5	3.60	46.7	27.8
70	31.25	4.08	41.2	24.5
80	25	4.59	36.6	21.8
90	18.75	5.07	33.1	19.7
100	12.5	5.48	30.7	18.2

(C) Measured Average Power with Program

Up to this point we have been discussing the values of average power that can be obtained using a static test signal rather than active video. Since TV Stations are in the business of transmitting pictures and not test signals, some value of PEAK POWER/AVERAGE POWER must be developed for use during modulation sequences.

We could assume that the value of the modulating video applied to the transmitter would be the mean between the 7.5 and 100 IRE unit limits when averaged over a long period of time but this was expected to vary considerably with program content. Therefore, the test setup shown in Figure 4 was developed to accurately measure, record, and calculate the APL Picture Power.

The measurement system consists of:

* Power Meter
* Digital Voltmeter
* HP - 75C Computer with peripheral equipment
* UHF TV Transmitter
* Directional Coupler

The power meter chart recorder output was connected to the DC input of the digital voltmeter. This permitted the direct measurement of the average power that was applied to the input of the power meter from the transmission line probe.

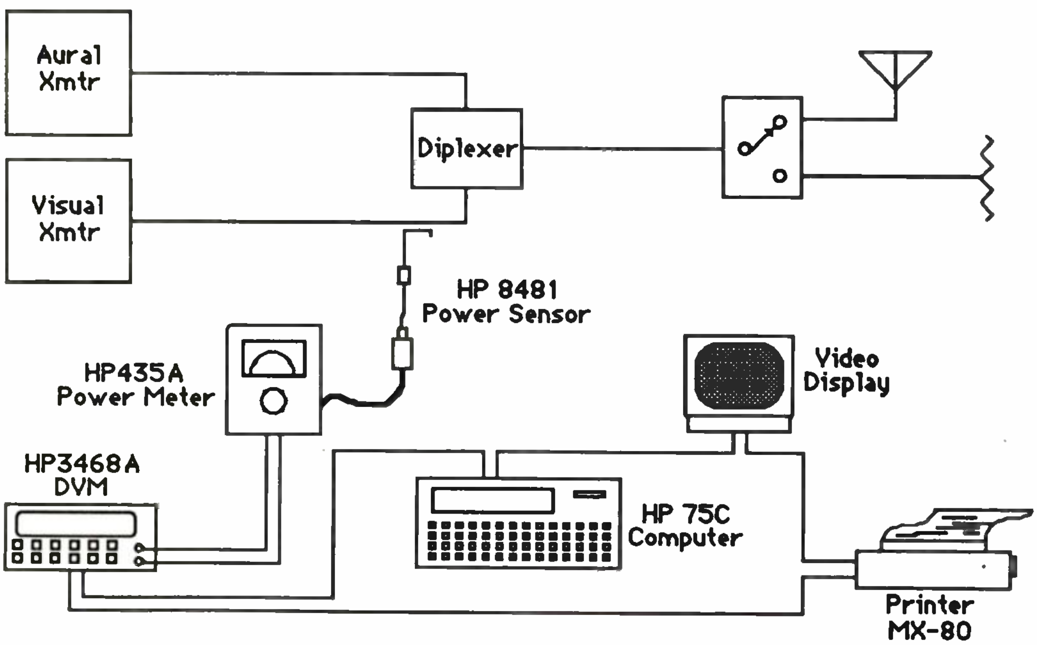


FIGURE 4

The combination powermeter and voltmeter was calibrated using the FCC prescribed sync and blanking signal and the calorimetric method. The transmitter was then modulated with a test signal consisting of sync and flat field modulation from 0 to 100 IRE units in 10 IRE unit steps. The individual DVM and average power values were recorded for each of the steps and tabulated in Table 3 .

Since the data was accumulated in DC volts, a constant had to be developed to convert the total DC value at the end of the test period into average power. The measured average power was divided by the measured DC voltage to achieve a constant for each 10 IRE unit step. The mean constant was calculated and stored in software to later convert the total DVM reading to APL Picture Power.

In our particular test arrangement, the combination of directional coupler and wattmeter resulted in a constant of .0292 .

The HP -75C computer was programmed to sample the voltmeter at a random rate and for a random period during the broadcast day. The computer interrogated the DVM to detect the presence of visual carrier and to shut down the measurement system during off-air periods. This was necessary as the accumulation of zero average power values, during these off-air periods, would have resulted in an error in the total data and samples at the end of the five day test period. The random sample period technique was employed (4) to insure that the sample data was of a truly random nature and not dependent upon any particular program.

Key data was also accumulated and printed to allow the calculation of the APL Picture Power or the maximum average for any given measurement period. A sample data page is shown in the Appendix.

TABLE 3

IRE UNITS	CALCULATED AVERAGE POWER	DVM READING (DC Volts)	CONSTANT
0	31.80	.895	.0281
10	27.90	.802	.0287
20	25.20	.717	.0284
30	21.35	.637	.0298
40	19.79	.563	.0284
50	17.11	.499	.0292
60	15.03	.444	.0295
70	13.03	.390	.0299
80	11.68	.348	.0298
90	10.53	.314	.0298
100	9.70	.291	.0300
MEAN	18.47	.536	.0292

Test Results

With the transmitter connected to the antenna with normal program content, the data was accumulated from February 26, 1983 to March 2, 1983. Table 4 lists information pertaining to the five day test period.

TABLE 4

DATE	PERIOD START	PERIOD END	TOTAL DATA	TOTAL SAMPLES	APL kW	SAMPLE PROGRAMMING
02/27/83	07:16:05	13:52:56	21,212	37,835	19.20	Religious/Ethnic/Movie
02/28/83	07:44:20	10:29:57	11,056	15,810	23.95	Religious/Entertainment
02/28/83	12:35:02	19:59:14	25,746	42,340	20.83	FNN/Educational/Entertainment
03/01/83	07:44:26	10:00:43	7,330	12,970	19.36	Public Affairs/Religious
03/02/83	07:44:27	09:59:52	7,010	12,900	18.61	News/Religious

All of the programming was aired in color during the test period with the exception of one half-hour of monochrome from 10:00 to 10:29:57 on February 28th.

The transmitter parameters in effect during the test period were:

Peak Visual Power	55 kW
Aural Power	5.5 kW
Call	WWHT TV
Frequency	Visual 795.25 MHz
	Aural 799.75 MHz
Channel	68

The results of the five day test period were:

Total Data Collected	72,354 DC Volts
Total Samples	121,855
Mean Value	.59377 DC Volts
Mean Average Power	20.33 kW

The mean DC voltage was obtained by dividing the total DC volts by the total number of samples taken.

$$\textbf{MEAN DC VOLTS} = \frac{\textbf{TOTAL DC VOLTAGE}}{\textbf{TOTAL SAMPLES}}$$

or

$$\textbf{.59377 DC VOLTS} = \frac{72{,}354}{121{,}855}$$

The Average Picture Power in kW was then calculated by dividing the mean DC volts by the constant .0292 that was derived for our particular test setup. This is the same procedure used during the calibration of the measurement system.

$$\text{AVERAGE PICTURE POWER} = \frac{\text{MEAN DC VOLTAGE}}{\text{CONSTANT}}$$

or

$$20.33 \text{ kW} = \frac{.59377 \text{ DC VOLTS}}{.0292}$$

Applications

The key application of the APL Picture Power is essential in:

(A) Determining the APL Picture Power radiated from a TV facility for predicting non-ionizing radiation levels

(B) Determining the true average power on RF components including transmission lines

(A)Non-Ionizing Radiation

When predicting or measuring power density, the APL Picture Power-ERP value should be used. This is especially true if confirming measurements are to be made with meters that sample the average signal.

The maximum radiated visual APL Picture Power-ERP is shown below.

TABLE 5

TYPE	VISUAL MAXIMUM ERP	APL PP - ERP
Lo VHF	100 kW	37.0 kW
Hi VHF	316 kW	117.0 kW
UHF	5,000 kW	1,851.9 kW

(B) Transmission Line

Coaxial transmission lines are average power limited. In a recent insurance claim involving a transmission line failure it was necessary to determine if the failure was caused by exceeding the average power rating of the line, and if not, what the safety factor was.

At the time of the failure, the station was operating at a licensed power of 93.8 kW peak visual and 10.4 kW aural with the line rated at 63 kW (5). However, it was shown with this measurement technique the average power was in fact operated with a 28% safety factor.

TABLE 6

	LICENSED POWER	BLANKING POWER	APL PICTURE POWER
Visual	**93.8 kW**	**55.8 kW**	**34.7 kW**
Aural	**10.4 kW**	**10.4 kW**	**10.4 kW**
	104.2 kW	**66.2 kW**	**45.1 kW**
Maximum line rating	**63 kW**	**63 kW**	**63 kW**
Safety factor	**Overrated**	**Overrated**	**28 %**

Conclusion

The average output power of a television transmitter varies with picture content. This paper outlined a technique to quantify average power during normal programming and found that, over the long term, average power is equal to licensed (or peak) power divided by 2.7 instead of the more traditional value of 1.68.

The APL Picture Power for WWHT during the test period was 20.33 kW or 4.32 dB less than the licensed peak power.

Appendix

1 Federal Communications Commission, *Rules and Regulations, Volume III*, March 1980, Section 73.663(b).

2

A = Average power

$$(C)^2 \left[\frac{0.075\,H}{H}\right]\left[\frac{525-6}{525}\right] +$$ Horizontal sync. including equalizing pulses

$$(C)^2 \left[\frac{H-[(.07H)(2)]}{H}\right]\left[\frac{6}{525}\right] +$$ Vertical sync.

$$(.75C)^2 \left[\frac{.175H-.075H}{H}\right]\left[\frac{525-18}{525}\right] +$$ Horizontal blanking

$$(.75C)^2 \left[\frac{H-[(.04H)(2)]}{H}\right]\left[\frac{12}{525}\right] +$$ Vertical blanking not including serrations

$$(.75C)^2 \left[\frac{(.07H)(2)}{H}\right]\left[\frac{6}{525}\right] +$$ Vertical serrations

$$d^2 C^2 \left[\frac{H-.175H}{H}\right]\left[\frac{22}{525}\right] +$$ VITS

$$b^2 C^2 \left[\frac{H-.175H}{H}\right]\left[\frac{525-40}{525}\right] +$$ Active video

Where:

- A = Average power
- C = Peak carrier amplitude
- H = Time from start of one line to start of next line
- b = Average percent modulation of the active portion of video lines 21 through 262 1/2
- d = Average percent modulation of the active portion of VITS lines 10 through 20

3 The Federal Communications Commission in its *Rules and Regulations, Volume III*, March 1980, Section 73.682(a)(13) and (17) specifies that modulation be maintained between 7.5 and 100 IRE units.

4 Roger C. Millikan, "The Magic of the Monte Carlo Method," *Byte Magazine*, Feb. 1983, pp. 371-373.

5 RCA, *Broadcast Transmission Line Equipment Catalog, TR.1101A*, (Camden, NJ: RCA Broadcast Systems), Figure 7.

THE 15KW KLYSTRODE™: AN EFFICIENT AIR COOLED DEVICE FOR TELEVISION BROADCAST SERVICE

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Introduction

The Klystrode and its RF circuit comprise a UHF amplifier. The device generates a grid driven density modulated beam similar to that of a tetrode. The output power is removed via a resonant cavity much like a klystron's. This type of structure permits efficient class B operation, independent input and output circuit tuning and long life resulting from its klystron like gun.

Poor efficiency is a double edge sword. One must pay the cost of purchasing extra power to operate the device [2] and one must pay the cost of removing the excess power as wasted heat. This paper will present the test results for a new efficient 15KW air cooled Klystrode. It will be shown that this device for UHF-TV not only reduces the cost of purchasing power but also reduces the need for a complex cooling system and its associated costs. The possibility of air cooling higher power Klystrodes is also discussed.

The Klystrode and Circuit

The X2254 15KW Klystrode in its associated experimental circuit is shown in Figure 1a. The tube, shown in figure 1b, is approximately 2 ft. in length with a weight of 36 lbs. The circuit is very similar in appearance to those for the 30KW and 60KW Klystrodes. From top to bottom the various

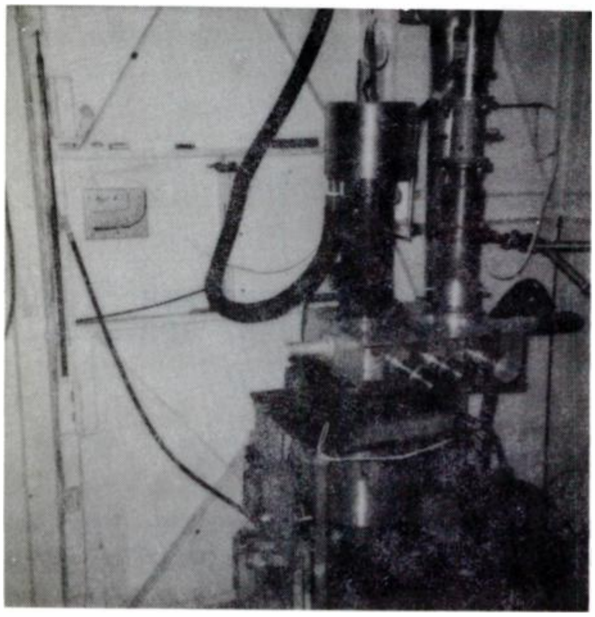

Figure 1a. X2254 in its UHF circuit.

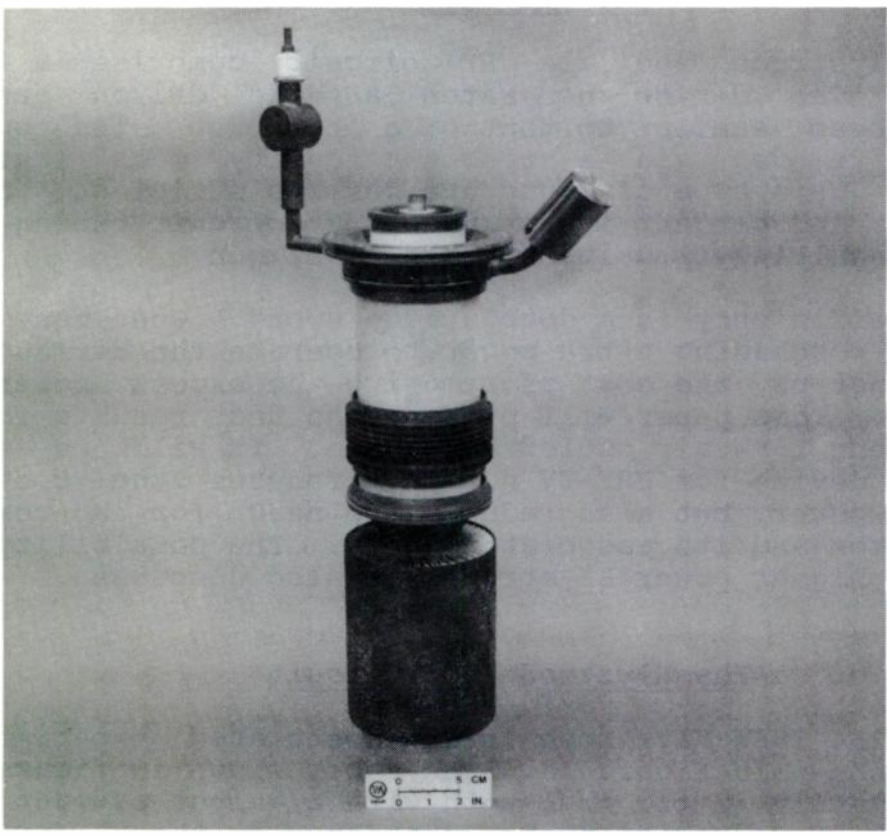

Figure 1b. X2254 air cooled Klystrode.

sections are: high voltage compartment, input circuit, output circuit and air-cooled collector circuit.

The high voltage circuitry is contained within a shielded compartment on top of the input circuit. This circuitry consists of various filters to contain the UHF fields and prevent instabilities at video frequencies. High voltage, grid bias and filament power enter this section via high voltage cables.

The input circuit is smaller in diameter than that used with previous Klystrodes. The drive is capacitively coupled to the input circuit through a small tunable coupling cavity (not shown). When properly adjusted, the coupling cavity and input circuit form a double tuned input circuit allowing wide bandwidth with reasonably high gain. Using Class B biasing, the RF driven grid much like a tetrode's density modulates an electron beam.

A single coil is used to generate the focusing field. An internal anode pole piece (see Figure 2) and external tail pipe (drift tube) pole piece shape the focusing field. For the data presented here, no magnetic return path was used. The Klystrode operates best with brouillon focusing. It can be assumed that a lower focusing power will be achieved with a proper return path. Less than 125 watts is expected.

The proper dimensioning of output gap and cavity were critical design parameters in obtaining the large required R/Q ratio. Without a large R/Q ratio it would have been impossible to achieve the required output bandwidth and good efficiency simultaneously. The output cavity is an iris-coupled double-tuned circuit. The output ceramic, 4.5 in. high and 4.0 in. O.D., is AD 99.5 and allows the use of the 60KW Klystrode output cavity, hence, a reduction in design and manufacturing costs. The generated power is extracted from the secondary cavity by a tunable capacitive probe type load coupler.

The tail pipe drift tube and collector are both air cooled. The tail pipe shown in Figure 3 was machined from a single piece of copper. However, future tail pipe fin assemblies may be brazed structures, similar to those used on present klystron drift tubes, to keep the manufacturing costs down. Very little air flow is required to cool the tail pipe drift tube.

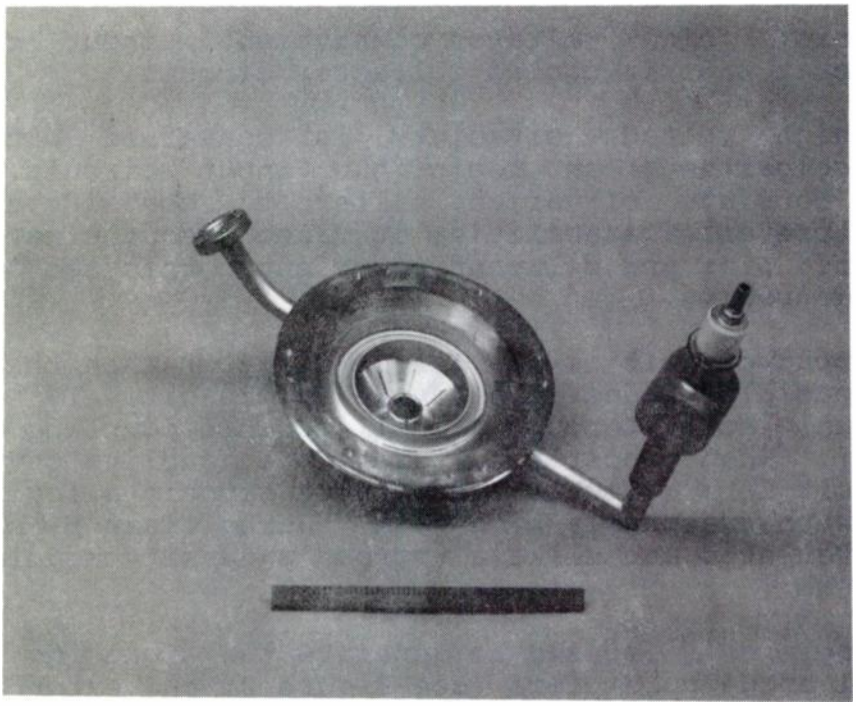

Figure 2. Anode and pole piece.

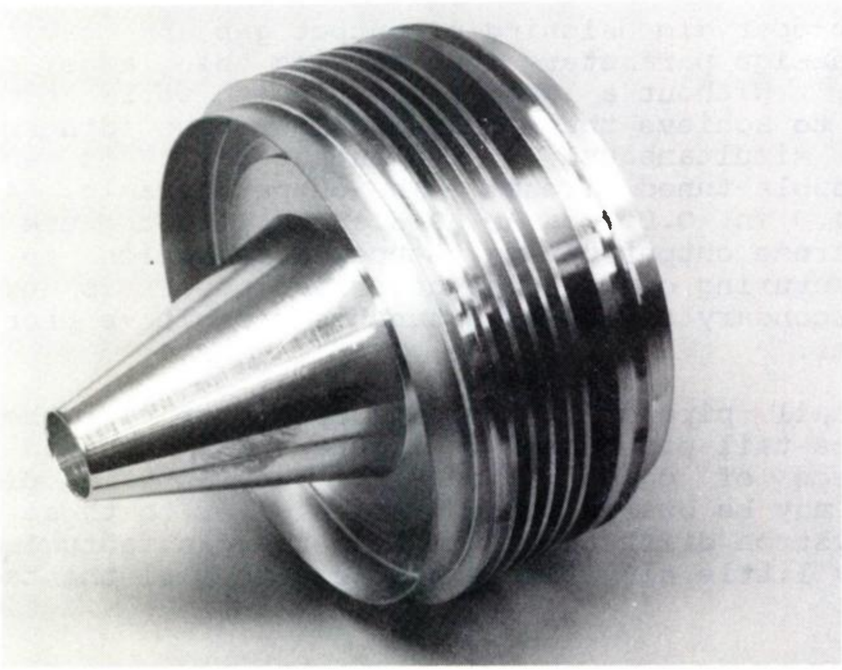

Figure 3. Klystrode tail-pipe drift tube.

X2254 Klystrode UHF Performance

The Klystrode program at EIMAC has been expanding for several years. Data has been published on visual and aural UHF Klystrodes at both the 30KW and 60KW levels [1,2,3,4]. The 60KW visual data has been independently been verified by Comark Communications [3]. The X2254 15 KW Klystrode is the latest addition to the growing family of devices.

Table 1 [5] lists the conversion efficiency for various CW output power levels. The output bandwidth for the given data is 9 MHz. With a device gain of greater than 22 dB, the drive power levels are considerably below 100 watts. A solid state amplifier capable of a peak power output of 90 watts and an average output power of 60 watts will easily drive an X2254 Klystrode in visual service. Such solid state amplifier modules are readily available. The output bandwidth for the data given is 9MHz.

Table 2 presents the low frequency linearity response of the X2254 calculated from the device's power transfer curve. Data is given for both conventional visual and multiplexed sound and visual operation. Although efficiency and output power are reduced, linearity is improved in the multiplexed mode. Sync compression is no worse than 51% for all modes of operation. The indicated degrees of nonlinearity are easily correctable with today's compensated drivers. Differential gain and differential phase were not measured. Their expected values were determined from those of the 60 KW Klystrode.

Table 1 X2254 UHF Performance Data

Frequency..........................772 MHz
Bandwidth............................9 MHz
Beam Voltage......................18.2 KV
Focusing Power....................240 Watts

Condition	Beam Current (A)	Power Output (KW)	Conversion Efficiency (%)	Drive Power (W)	Gain (dB)
Peak of Sync	1.59	15.5	53	86	22.6
	1.21	9.8	45	56	22.4
Black Level	1.10	8.6	43	50	22.3
Ave.Picture Level	0.66	3.4	28	38	20.8

Table 2 X2254 UHF Television Data

Operational Mode	Conventional	Multiplexed	Multiplexed
Peak of Sync Power (KW)	15.5	8.0	4.0
Quiescent Current (A)	.020	.67	.67
Nonlinearity (%)	-61	-48	-19
Sync Compression (%)	-41	-51	-26

Differential Gain..................-40% *
Differential Phase.................-8 Deg *

* Expected Value

Klystrode Efficiency

The key to energy efficient transmitter operation is to keep unnecessary power out of the electron beam. The conventional klystron is a Class A device. If one assumes a beam energy extraction efficiency of approximately 50% at peak of sync, this requires the power supply to deliver about twice peak of sync power to the beam. Since beam current and anode voltage are both constant in a conventional klystron, so will be the required DC supply power even if the output is decreased to white level. If mod-anode pulsing is used the situation is somewhat better. The DC beam input power will be somewhat greater than twice black level for all levels of output. All energy which is not transmitted ends up as waste heat in the collector.

The most efficient method of operation is to have the DC beam energy track the current picture level (or drive power level). The klystrode amplifier accomplishes this by operating in a Class B mode. The grid, which is RF driven, density modulates the beam current in proportion to the drive power level. This results in a DC beam input power which varies with the current picture level. The average DC input power will be slightly greater than twice the average picture level output. Hence, the energy cost of operation will be much less for a klystrode than for a conventional or a mod-anode pulsed klystron.

Another method of beam control is known as Full-Time Modulation (FTM). Here the beam current level is controlled at the cathode with the base bandvideo information. The RF is then added at the first cavity. Using this technique the beam current and DC input power will track the picture level. Although more efficient than mod-anode pulsing, FTM has its own unique problems. First is the difficulty of applying low frequency information to the modulating grid. This electrode is at or close to the high voltage cathode potential. Therefore, the wideband video amplifiers required to drive the grid must be isolated from the tube or floated and isolated from the source equipment. This is not an easy task at baseband. The second FTM problem relates to the generation of unwanted sidebands. FTM first imparts to the beam current the baseband video spectrum. The process of then velocity modulating the beam with RF is equivalent to a time domain multiplication or frequency domain convolution of the video spectrum and a carrier frequency. The result is a beam current which contains a carrier frequency and both the upper and lower sidebands. The lower can only be removed by elaborate filtering techniques at the output cavity. Besides being an added complexity, the removed sideband power is lost energy.

Neither of these problems occur in klystrode amplifiers. Its grid is RF driven. The input RF is coupled to the grid via a ressonant input cavity using a simple capacitive probe. The klystrode's drive, having been previously modulated and filtered, contains the desired upper and vestigial sideband information only. The process of modulating the beam with this drive is equivalent to the product of a constant and the desired RF spectrum. The results contain no extraneous sideband information.

Earlier articals [1,2,4] have discussed the importance of figure of merit (previously misnomered as peak of sync efficiency) as a parameter for comparing device efficiency and cost of operation. Figure of merit is defined as the ratio of peak of sync power output to the DC power input required for an average picture output. As this ratio increases the device and method of operation becomes more efficient (with an absolute maximum of about 300% corrisponding to 100% efficiency). The X2254 15KW Klystrode has a figure of merit equal to 103% compared to the typical klystron's value of 41% [2].

Efficiency and Air Cooling

One may ask, who cares about efficiency at 15KW? Collector cooling in any device is critical as a large fraction of the electron beam energy is dissipated in the collector as waste heat. The klystrode, being an efficient Class B device, has a distinct advantage over conventional klystrons and other Class A devices in that the average power disspation in the collector is much less. Since the beam energy in the klystron is constant at about twice peak of sync, at white level when the RF component of the beam current is much less than its DC content, almost all of the initial beam energy is dissipated in the collector. In contrast, the klystrode's beam energy varies directly with drive level. Therefore, at white level there is very little energy in the beam and very little power is wasted in the collector. Hence, efficiency relaxes the cooling requirements and costs. The klystron's worse case is a prolonged white level ; the klystrode's is a prolonged black level.

The X2254 has introduced the concept of total air cooling to the klystrode family. As with previous Klystrodes, both the cathode circuit and output cavity are air cooled. In addition, the collector and tail pipe drift tube are air cooled. This simplifies the cooling system by allowing one blower to cool the entire circuit.

The X2254 collector and drift tube surface temperatures are listed in Table 3 [5] as a function of output power. It should be noted that these temperatures drop as the output power drops. A klystron with a similar collector and cooling system might start out at equal or slightly higher surface temperatures when running at peak of sync. However, as output power is decreased the collector surface temperature would rise

Table 3 Collector Surface Temperature Vs. Output Power*

Power Output (KW)	Tcb (C)	Tct (C)	Ttp (C)
15.5	202	132	90
9.8	160	110	85
8.6	158	115	77
3.4	140	88	60

* Focusing Power = 240 Watts

All of this means that a klystron collector must be capable of dissipating an average power of more than twice peak of sync power. However, a Klystrode collector only needs to dissipate an average power of slightly greater than black level. The decrease in average collector dissipation results in: a smaller collector fin structure, a lower air flow and a lower pressure drop across the fin structure. Therefore, the Klystrode affords a more cost effective and quieter cooling system.

With these advantages in mind, an inexpensive collector has been designed (see Figure 4). The structure consists of a bored copper cylinder. Wrapped around and brazed to the outside of the cylinder is a corrugated copper fin structure with an outer sheathing. The cylinder O.D. is 2.5 in. The fins are 6.0 in. long and 1.25 in. wide for a total collector O.D. of 5.0 in. The top of the collector is flared to allow a smooth directional change in air flow.

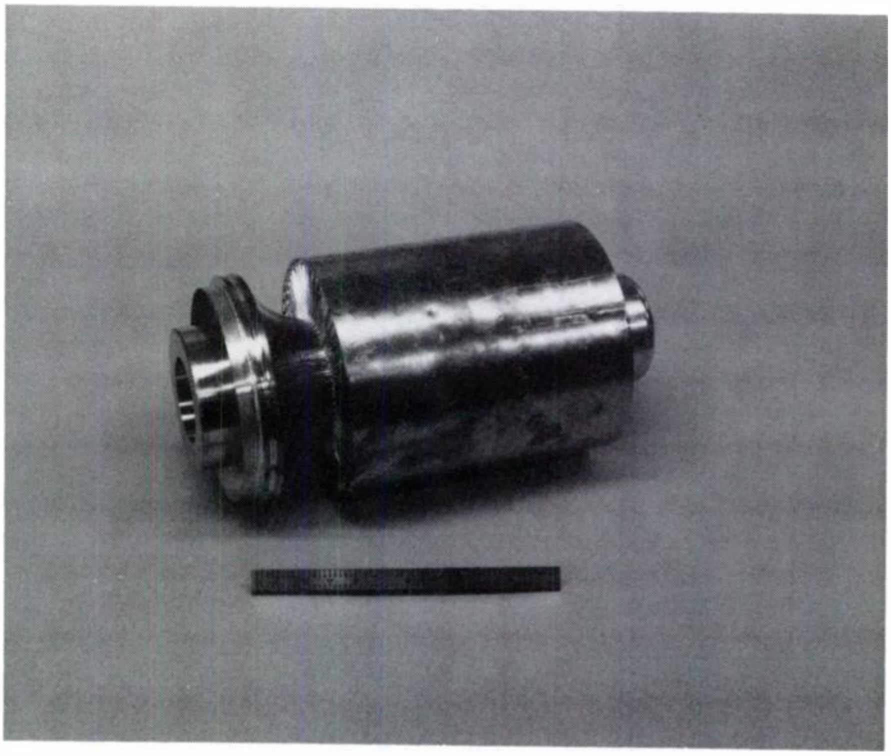

Figure 4. Collector and fin assembly.

For the data presented here, air was forced from the bottom of the collector back toward the collector's mouth and exhausted through a ducting system as shown in Figure 5. Flow could also have proceeded in the opposite sense. Thermocouples were placed on the cylinder surface at the top and bottom of the fin structure. Figure 6 [5], the collector characteristics at slightly greater than black level, show pressure drop and surface temperature as a function of air flow. The curves indicate that collector surface temperatures remain manageable even down to very low air flow rates. By varying the magnetic field strength, the position of beam interception on the collector wall can be shifted. This can be observed by a transposition of the two temperature curves for a low enough field.

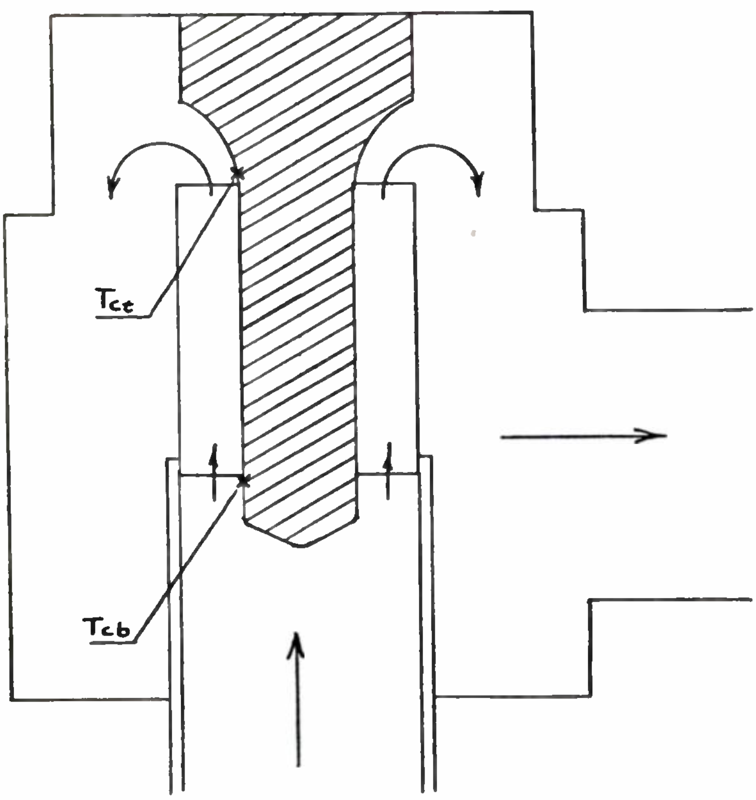

Figure 5. Experimental plenum and thermocouple positions.

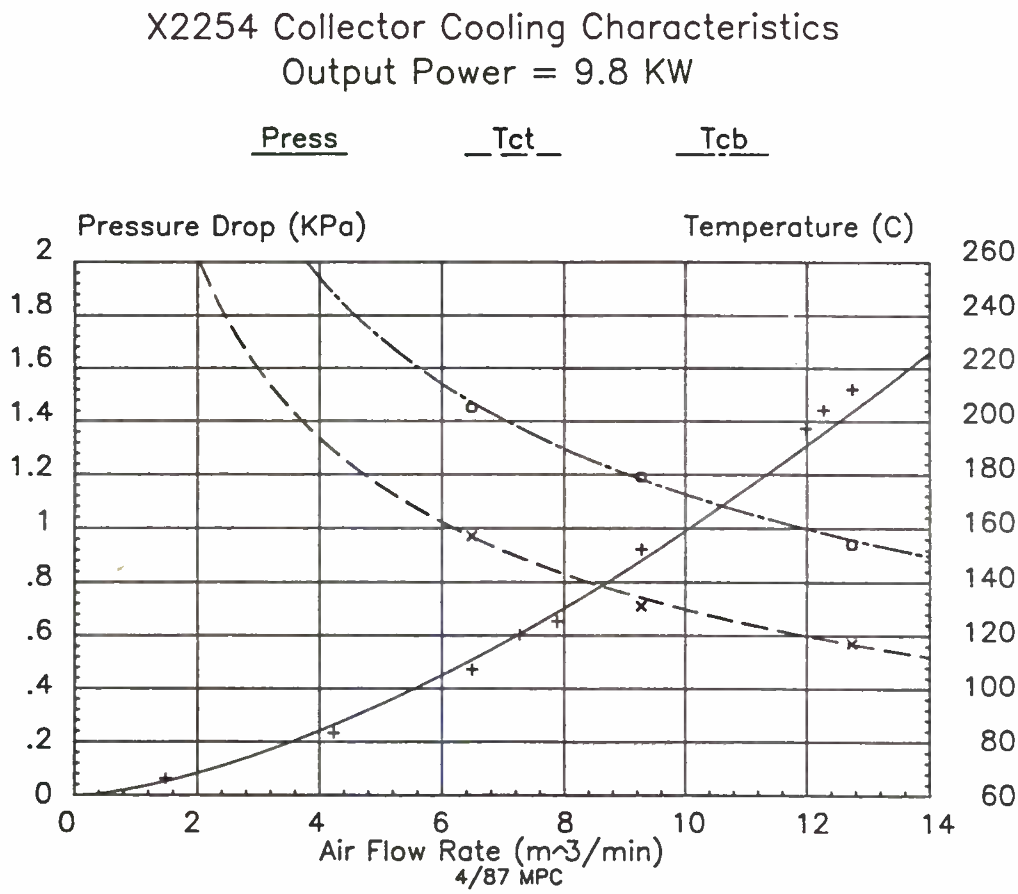

Figure 6.

Future Klystrode Devices

The small collector size, low pressure drop and low air flow required to cool the 15KW Klystrode suggests that air cooling may be a viable technique at higher power levels. Air cooled 15KW klystrons presently exist. Since the average collector dissipation in a klystron is much greater than twice that of a comparable Klystrode, a 40KW air cooled Klystrode could be designed using the 15KW klystron's collector. Also, designing a 60KW air cooled Klystrode now becomes practical.

A Note About Klystrode Life

Up to the present there has been much speculation in the industry as to the life of a Klystrode. The main mechanism of failure in tetrodes and klystrons is cathode life. The life of a dispenser cathode is dependent on required cathode loading or temperature only. The typical klystron cathode lasts 40 Kh or longer. Gridded klystron gun structures have been reported to last in excess of 35 Kh. It should be pointed out at this time that the gun structure of a Klystrode is very similar to that of a klystron. In fact, the dispenser cathode is also identical to those used in many existing UHF klystrons. Hence, 30 Kh would then be a conservative estimate of Klystrode life. EIMAC has logged 5Kh at 5KW (a power slightly greater than average picture level for the 15KW Klystrode) with no ill effects. In the field operation will be the only true test of Klystrode life. This should occur some time in 1988.

Concluding Remarks

Previous papers [1,2,3,4] have shown the advantages of using the Klystrode in UHF television service at higher power levels. It was the intention of this paper to present information on the new EIMAC X2254 15KW air cooled Klystrode. Data was presented showing the low frequency linearity and sync compression both of which are correctable. Also given was the maximum expected differential gain and differential phase. It was shown that the X2254 is energy efficient which allows it to use a simple, quit and economical cooling system. This implies the possibility of extending the technology of air cooling to higher power Klystrodes without having to use a blower of extreme horsepower. Finally, the life of the Klystrode should easily approach that of a klystron.

Bibliography

[1] Preist, Donald H. and Shrader, Merrald B., "Recent Developments in Klystrode Technology", Varian EIMAC, 14th International TV Symposium, Montreux, Switzerland, symposium Record, Broadcast Sessions, 1985.

[2] Badger, George M., "The Klystrode, A New High-Efficiency UHF TV Power Amplifier", Varian EIMAC, Proceedings of the National Association of Broadcasters 63rd Annual Convention and International Exposition, 1985.

[3] Ostroff, N. and Whiteside, A., "Using Klystrode Technology to Creat a New Generation of High Efficiency UHF-TV Transmitters", Comark Communications, Proceedings of the National Association of Broadcasters 64th Annual Convention and International Exposition, 1986.

[4] Preist, Donald H. and Shrader, Merrald B., "The Klystrode-An Unusual Transmitting Tube With Potential For UHF-TV", Proceedings of the IEEE, Vol. 70, No. 11, Nov. 1982.

[5] Chase, Michael P. and Preist, Donald H., "A New 15KW Air Cooled Klystrode", Varian EIMAC, 15th International TV Symposium, Montreux, Switzerland, Symposium Record, Broadcast Sessions, 1987.

GATED IMPULSE MODULATION

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ABSTRACT

Gated Impulse Modulation (GIM) is a new electronic communication method combining message and clock information into a single digital train for amplification, transmission, storage or processing. Developed by Focus Communications, Inc., GIM is generated at logic level, yielding a signal compatible with computers, data channels and a host of other digital equipment.

A GIM signal has the unique characteristic of being convertable simply and directly to a standard amplitude modulation envelope, with information modulating clock-based carrier content. A GIM signal containing audio and 1 MHz timing, for example, may be converted by passive means into a 1 MHz amplitude modulated carrier. Since a digital GIM waveform contains all necessary a.f. and r.f. information, it can be amplified in high efficiency Class D (switch mode) stages to any desired level - 1 watt, 50,000 watts, etc. - before conversion to standard AM. One result is AM transmitter power efficiency previously unimaginable. For example, a typical AM broadcast transmitter in use today might achieve an overall efficiency of 45%, requiring 22,000 watts of mains power to send 10,000 watts of modulated carrier to the antenna. In contrast, a GIM transmitter might attain 90% efficiency, needing only 11,000 watts to output 10,000.

Such high operating efficiency is attained because the entire signal, including carrier and message information, is amplified in a single channel, eliminating the separate modulator which normally comprises 50% of an AM transmitter. Moreover, since all amplification takes place in Class D, very little power is dissipated as heat. The resulting GIM transmitter could utilize half the components, take up half the space, and require half the spare parts and maintenance time of current models.

Because of its binary nature, a transmission system employing GIM can accept digital outputs from CD, DAT, digital carts and the like, process these signals digital/digital (D/D) in a GIM converter, then amplify the resultant to the desired power level prior to AM envelope restoration. Consequently, the advantages of digital audio - fidelity, immunity from hum, distortion and noise - are preserved all the way to the output network, overcoming a major obstacle in the battle to make AM transmitters competitive with FM.

The remainder of this paper will explain how Gated Impulse Modulation is achieved, then focus on some of its uses and advantages.

FUNCTIONAL ANALYSIS OF GIM

Figure 1 depicts functional block diagrams of two AM transmission systems employing Gated Impulse Modulation, and Figure 2 illustrates corresponding waveforms at various points in Figure 1. Referring to Figure 1 A, analog audio (waveform 1) enters an analog-to-digital (A/D) converter which changes the a.f. signal into a Pulse Duration Modulation (PDM) train like that depicted in waveform (2). Note that the 100 microsecond interval shown in waveform (2) corresponds to the downslope segment marked in waveform (1). Thus it can be seen that the PDM duty cycle varies from almost fully positive (or "on") at the high point of the a.f. waveform, through 50% on/off in the middle region where a.f. crosses zero, to nearly completely off at the end corresponding to the most negative region of the incoming audio.

For the sake of clarity, a relatively high frequency 5 kHz tone is shown entering at (1) so that a true idea of the PDM sampling rate can be given in waveform (2). In practice, PDM sampling takes place at an odd fraction of the r.f. carrier frequency, f_c. For example, a PDM frequency of $5f_c/32$ means that audio will be sampled every 6.4 r.f. cycles. Such a rate yields a PDM frequency of 156.25 kHz for a station on 1000 kHz, the center of the medium wave band. A PDM rate at 560 kHz would still be 87.5 kHz, sufficient to sample 10 kHz audio 8.75 times per cycle.

The clock unit shown in Figures 1A and 1B provides PDM timing signals to the A/D converter, the same in both diagrams. The clock also sends carrier frequency information to a GIM former, which passes carrier waveform (4) only when gated by PDM signal (2). The GIM former may utilize AND or NOR circuitry to achieve this. Shown in waveform (5) is a 32 microsecond portion of the GIM signal thus formed, in this case by ANDing carrier information (4) with a PDM representation (2) of incoming audio (1). The interval shown in waveform (5) corresponds to the first third of waveform (2), whereas waveform (5A) corresponds to the entire 100 microseconds of waveform (2).

Note that only when the PDM signal is in an "on" mode does carrier information gate through the GIM former. In the absence of an audio signal, zero modulation would be signified by a 50% on/off PDM duty cycle, and carrier information is gated through in 3.2 cycle bursts like those shown in the middle part of waveform (5A). To best understand how a full AM carrier is restored, let us consider this unmodulated case first.

After the Gated Impulse Modulation former, a logic level digital signal like that shown in waveforms (5) and (5A) containing both carrier and message information is passed to any number of Class D, switch mode amplifiers for power amplification to some desired level: 5 watts, 500 watts or 50 kilowatts. Since the GIM signal is binary in nature, it can be amplified at power out/power in efficiencies approaching 95%.

Whatever the power level, the waveform leaving the final amplifier stage is essentially the same digital train that left the GIM former (waveform 5), containing bursts of carrier information in proportion to the instantaneous duty cycle of a PDM signal which in turn represents incoming audio or other information. Thus amplified, the GIM train is ready to enter the final transmitter stage, a passive AM-restoring tank circuit consisting of input coupling capacitor, a high-Q L or LC shunt circuit tuned to resonate at exactly f_c (1000 kHz in the example), a trap circuit at f_{PDM}, and other more conventional PA output-to-transmission line tuning elements.

The AM restorer actually serves five functions: first, it supports signal excursions at the carrier frequency, storing and releasing energy to fill in the "holes" between bursts of r.f. provided by the GIM signal. In this manner it acts in "flywheel" fashion, outputting a full r.f. carrier from the 50% duty cycle (unmodulated) signal it receives. The second function of the AM restorer is to eliminate PDM switching artifacts. This is accomplished largely in the high-Q circuit resonant at f_c, which does nothing to support PDM frequencies occurring at 5/32 the resonant frequency. Here it can be seen that PDM switching at some even fraction of f_c (e.g., 1/5 or 1/6) would be much harder to remove from the AM envelope. The 1-2% f_{PDM} artifacts remaining after passing through the resonant tank are easily removed by an LC trap tuned to the PDM frequency.

The third function of the AM restorer is to change the carrier waveform from digital to sinusoidal; that this is a task easily performed is evidenced by the many solid state broadcast transmitters now accomplishing the same in their final tank circuits. In fact, the high-Q network immediately turns the square wave signals into sine waves, and the process is enhanced in succeeding stages of the tank. Fourth, the restorer serves the conventional functions of harmonic removal and as matching network from final power amplifier stages to transmission line.

Finally, the restorer acts as a 10 kHz low pass filter to assure a smooth, relatively slow transition between high and low carrier levels as represented by modulation peaks and valleys, exactly reproducing the excursions of the original analog modulating information without passing PDM actifacts. To understand how this is accomplished, let us consider a case where an incoming audio signal is converted to PDM (waveform 2) to effect 100% modulation of the restored AM envelope (waveform (6). At its full positive peak, the audio generates a PDM waveform mostly "on" as depicted at the beginning of waveform (2). This signal enters the GIM former, gating 100% of carrier through at maximum "on" PDM, and only 50% when no audio is present (waveform 5).

As the audio signal goes negative, PDM "on" time is decreased until, finally, it is not "on" at all at maximum audio negative excursion (end, waveform 2). This results in less and less carrier passing through the GIM gate, as shown at the right end of waveform (5A). Let us slow the modulation rate down to almost zero, entirely possible in a GIM transmitter, which can faithfully reproduce D.C. differences entering the A/D converter: at zero D.C., we have a certain value carrier ouputting the AM restorer, measuring, for example, 500 volts peak-peak. As the D.C. voltage goes below zero, the PDM waveform has proportionatly less and less "on" time, and less and less carrier energy is passed to the AM restoring network. As this occurs, the transmitter will put out ever less r.f. energy, until finally it will cease to output anything at all (zero volts peak-peak). This point corresponds to a zero "on" time for PDM, and sets the limit for 100% negative modulation.

Increasing the D.C. level entering the A/D converter above zero will correspondingly increase PDM "on" time and allow more carrier energy to pass to the AM restorer. This will result in an increase in carrier level: to 600 volts, 700 volts, and finally to 1000 volts peak-peak. This point corresponds to 100% PDM "on" time, and represents 100% positive modulation. The + D.C. level entering the A/D converter will be exactly the same as the - level which resulted in zero PDM "on" time and zero carrier out. For a higher level of peak positive modulation (e.g., 125%), the A/D reference voltage may be adjusted so that no modulation results in a 40% PDM "on" time, 100% negative audio in 0% PDM, 100% positive audio in 80% PDM, and 125% positive audio yields 100% PDM.

Returning to Figures 1A and 1B, there is shown two D/D converters which accept digital audio from program sources such as compact discs, digital tapes and digital carts (waveform 3). This digital/digital converter employs microprocessing to reformat the incoming signal, which is typically in Pulse Code Modulation (PCM) form, to either a PDM (Figure 1A) or GIM (Figure 1B) format synchronized to the transmitter clock. In the first case, the GIM former selects between incoming PDM signals, formed either from analog or digital inputs, while in the latter (Figure 1B), selection is made at the input of the power amplifier train between GIM signals formed from either analog or digital programming.

OPERATIONAL RESULTS

A number of prototype Gated Impulse Modulation circuits have been made in Focus Communications laboratories, including low power transmitters in the under-five watt range which have been test broadcast. Results have been most encouraging, with total distortion less than 1%, and audio frequency response flat from D.C. to 15 kHz. A model 2 watt transmitter employs 5 standard IC's and 2 transistors; a 200 mW unit uses an audio transistor, 4 shelf CMOS IC's and takes about $15 in parts. A GIM card containing a broadcast-stable master clock, f_c and f_{PDM} timing, analog/PDM conversion, variable gain audio amplification and GIM former outputting TTL or CMOS wavetrains can fit in the palm of the hand, drawing 30 milliamps from a single-ended 12-volt supply. It is our hope to have a full-size transmitter employing GIM approved and on the air somewhere in the next 12 months.

THE FUTURE

Even as AM radio flounders in the United States and other countries, new broadcast technologies are coming to the fore to assure the viability of "standard broadcast." In the months and years ahead, we shall see pre-emphasis, 10 kHz bandstop, r.f. masking, low-angle antennas and AM stereo adopted by more and more medium wave stations, sending receiver manufacturers the signal that AM broadcasters have gotten down to business and intend to stay that way. Among these new technologies, one of the most exciting developments is digital audio: already, 17.4% of the radio broadcasters in the U.S. are using compact disc (CD) players on the air. But the figure dims for AM stations, only 7.7% of whom employ CD, as compared with 23.8% for FM users.[1]

The reasons for the disparity are several. Most AM stations are not doing too well at present, capturing only 28% of the radio audience as opposed to FM's 72%.[2] Therefore, they may not be in a position to acquire new equipment as rapidly as their FM counterparts. A second reason is more subtle: AM is perceived in many quarters - and even by broadcasters themselves - as an inherently "inferior" medium technically. So why throw good money after bad? goes the argument. This point of view is not only defeatest, but unfortunate as well in light of technology that can and will come to AM's rescue.

To return to the digital audio case, it is likely true that in their present condition, many AM stations would receive little technical quality improvement by using CD's and other digital program sources on the air. In fact, in <u>Gazing Into The Crystal Ball</u>, the NAB's "Radio Station Manager's Technological Guide To The Future" (1987) we read:

"Radio stations must be cognizant of the way listeners <u>perceive</u> their sound. As it turns out, listeners will largely hear what they <u>expect</u> to hear....For example, if you promote the fact that digital audio is superior to analog, and that your station has a 'CD Hour,' listeners may well believe that they hear a better sound, whether or not the engineers can objectively measure the qualitative difference in the received audio signal."

There may be some truth in the above statement, but a far better justification for "going digital" involves the very increase in quality - measurable and perceivable - that its writers seem to doubt. To prepare an AM station for an "FM-like sound," work must be done from the antenna backwards. One big hurdle is the transmitter, often a 20- or 30-year old relic which has been "peaked up" to maintain signal strength and increase overall efficiency 10%. This may keep the station at its legal power level and save something on the electric bill, but more often than not such "peaking" results in higher transmitter Q, with proportionately narrower bandpass. If this is your situation, of course a CD won't sound any different on the air!

Moreover, transmitters and studio equipment alike introduce other problems between program and antenna: hum, distortion, hiss and static, oscillations, poor low or high frequency response, sluggish slew rates, etc., ad nauseam. In an all-digital AM station featuring CD, DAT and other program sources not **re-converted to analog, but instead directly inputting a digital transmitter employing GIM or a similar technique, all the above problems are avoidable.**

I would predict that the next decade will see the advent of all-digital AM radio. Probably the first users will be high power stations who may be able to save thousands of dollars monthly in electric consumption, and small community stations, for a scaled-down version of much the same reason. In fact, one projected use for Gated Impulse Modulation is in Village Radio, a low power (e.g., 1-100 watt) informal broadcast concept utilizing a few thousand dollars worth of equipment to serve the needs of a small and often remote community. Village Radio stations in many areas of the world might be energy self-sufficient, operating from storage batteries charged by solar energy. An application for just such a station is pending in one country, and several more are in preparation.[3]

Here are a few more predictions for the future of AM radio, taken from a paper by the same name being presented at the International Sysmposium on Broadcasting Technology in Beijing, China (Sept 24-26, 1987):

"AM radio need not disappear; indeed, it will not disappear. The medium has a transmit-receive quality potential nearly rivaling FM, and it utilizes less than 1/10 the bandwidth to achieve this. AM represents too efficient a transmission method for governments to be apt to discard....

"The decline of AM vs. FM audience shares in the U.S. and elsewhere will gradually diminish, halting a couple of years hence with AM's share at around 15%....The broadcasters inheriting AM radio will set about to improve it in every manner....

"Regulatory agencies will take a role in the "cleanup" of the medium wave AM broadcast spectrum, granting licenses for repeaters in shadow areas, discouraging interference, and assisting in the release of new and needed technologies....

"Inexpensive "village radio" will become commonplace in various parts of the world. Many village stations will be energy independent, relying on photovoltaic cells or other alternate energy forms to charge batteries to power the transmitter, studio equipment, lighting, ventilation and other needs....All-digital AM broadcast will arrive and change once and for all the tarnished quality image of this medium. Receiver manufacturers will take note, and AM radio will take its rightful place alongside FM."[4]

It is my hope that Gated Impulse Modulation may contribute in some part to the restoration of AM radio as the most-utilized mass medium in the world.

References:

1. John D. Abel and Richard V. Ducey, "Gazing Into The Crystal Ball," National Association of Broadcasters, Washington, D.C., January 1987.

2. Editorial, "Same Song, Umpteenth Verse," Broadcast Engineering, Overland Park, Kansas, June 1986.

3. The author has several articles pending publication on this subject. For more information, contact him directly.

4. K. Dean Stephens, "The Future of AM Radio," 1987 International Symposium on Broadcasting Technology, Beijing, China, September 1987.

FIG. 1A

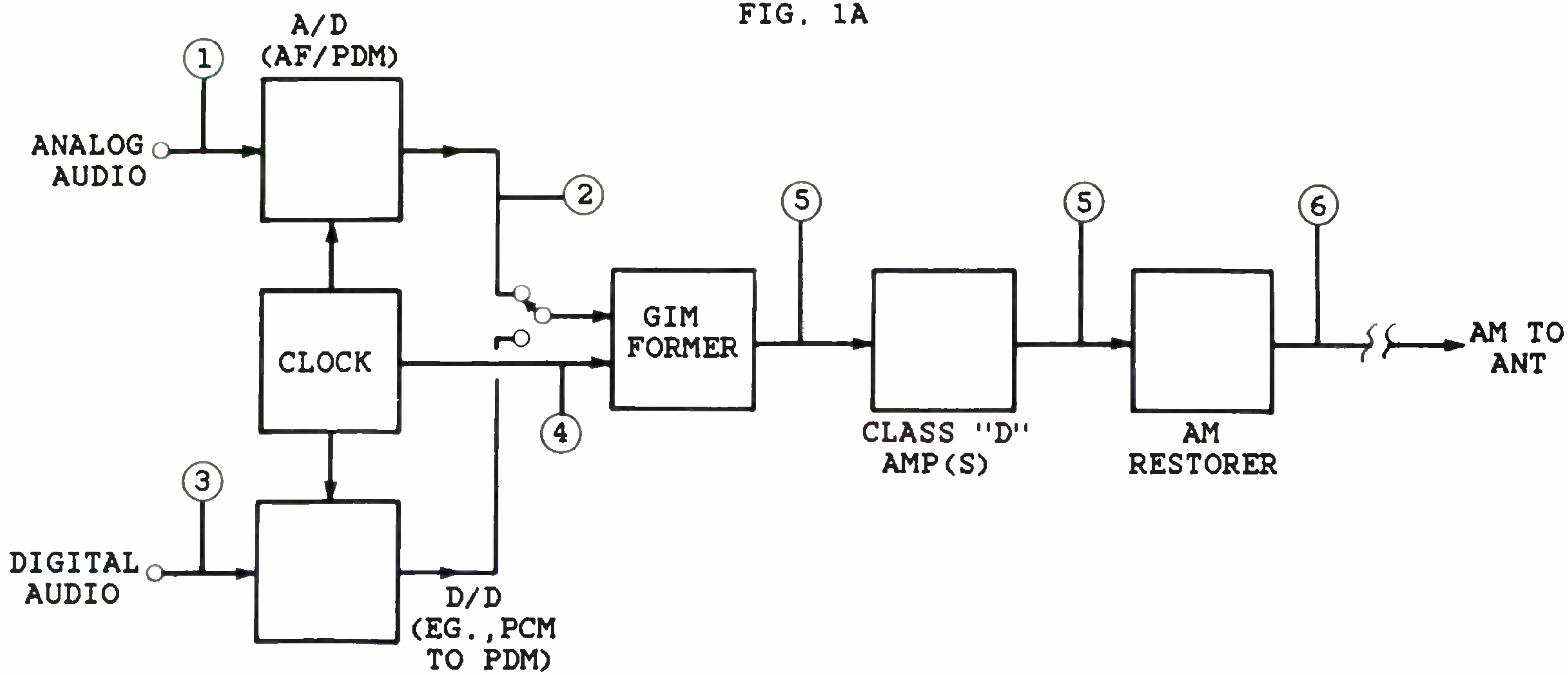

A/D
(AF/PDM)
1
ANALOG AUDIO
2
CLOCK
GIM FORMER
5
5
6
AM TO ANT
CLASS "D" AMP(S)
AM RESTORER
4
3
DIGITAL AUDIO
D/D
(EG.,PCM TO PDM)

FIG. 1B

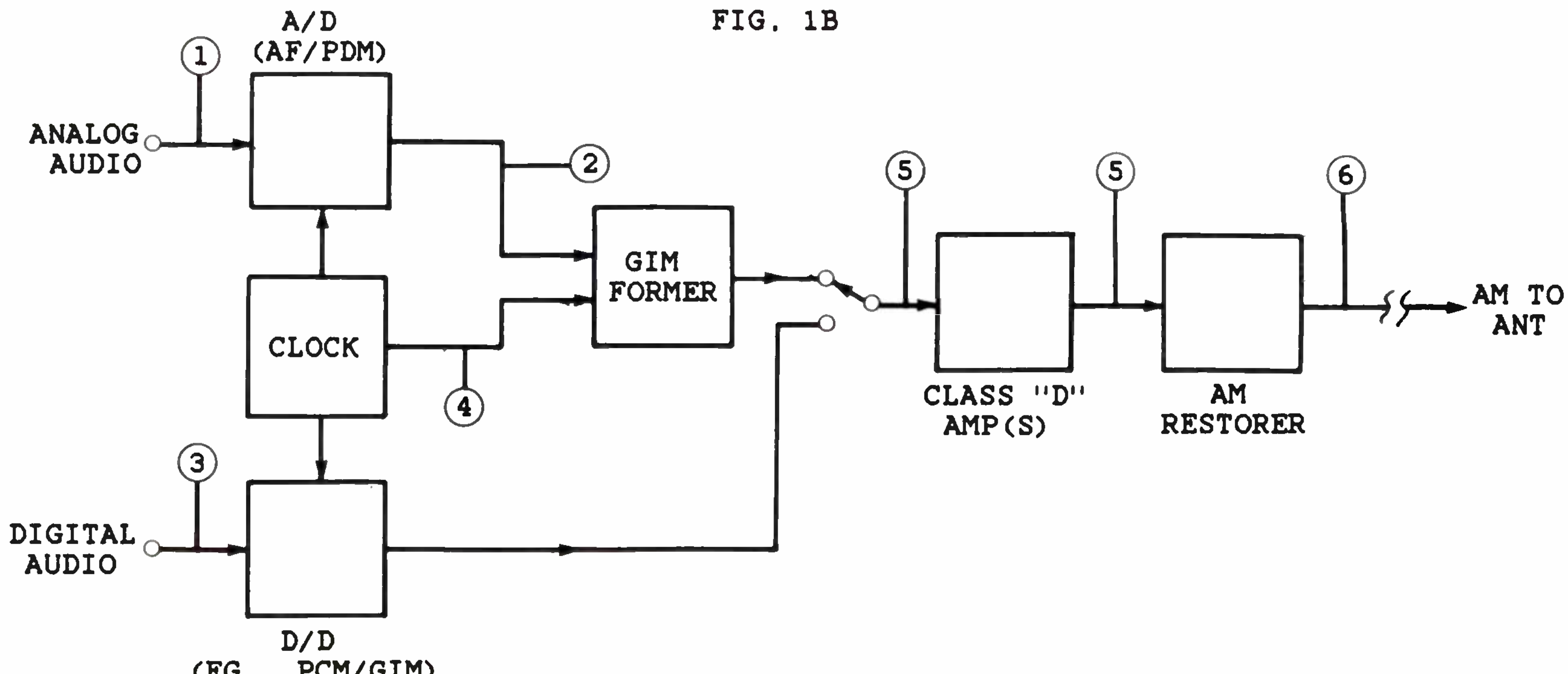
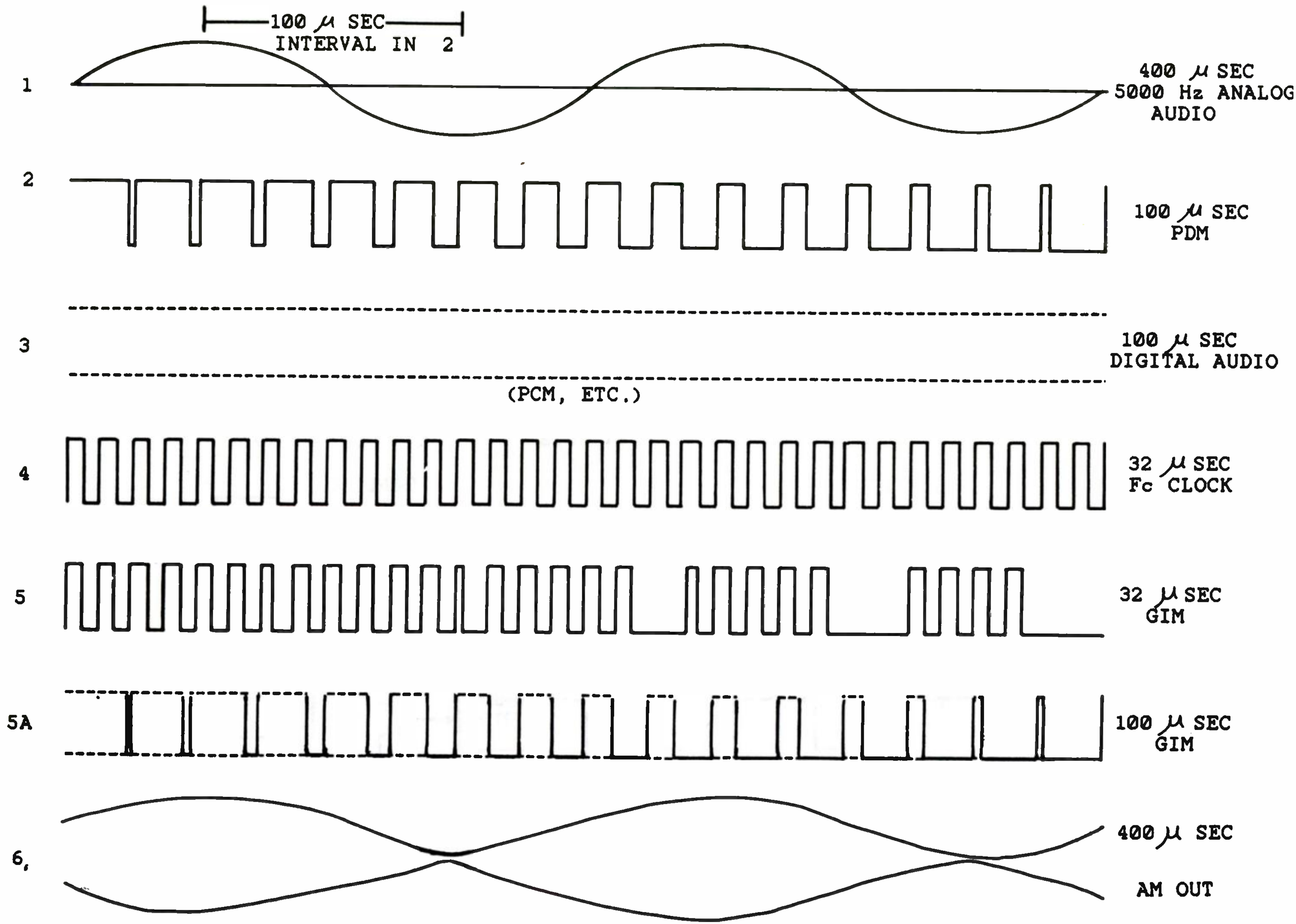

A/D
(AF/PDM)
1
ANALOG AUDIO
2
GIM FORMER
CLOCK
4
5
5
6
CLASS "D" AMP(S)
AM RESTORER
AM TO ANT
3
DIGITAL AUDIO
D/D
(EG., PCM/GIM)

FIG. 2
GATED IMPULSE MODULATION - WAVEFORMS



FLATTERER™ - A NEW CONCEPT FOR BROADBANDING AM ANTENNAS

by Leonard R. Kahn

President, Kahn Communications, Inc.
Westbury, New York

Introduction

AM broadcasting, in order to effectively compete with more modern means of delivering high fidelity signals, must update its entire plant. It is only too obvious that the weakest link in the overall system is the AM receiver. Hopefully, competitive marketplace pressure from high fidelity receiver manufacturers, supported by on-the-air promotions of quality AM radio, will force mass producers to improve the performance of their AM radios.

On the transmission end, most equipment used by broadcasters is of excellent quality. However, there is, and has been since the very beginning of broadcasting, a weakness in the transmission end of AM broadcasting. This is the limited bandwidth of the antenna system which, in addition to being an efficient radiator, must, for many stations, provide substantial directivity.

AM broadcasters, especially those operating stations on the low end of the dial, find it most difficult to transmit high fidelity signals because the percentage bandwidth of their antennas are quite substantial. Sophisticated methods have been developed for broadbanding such antennas, but implementation of some of these designs is expensive, and critical to adjust. On the other hand, the author would like to emphasize the fact that most stations will find the conventional broadbanding approach quite satisfactory and this new technique will be of most interest to high powered stations operating on the low end of the band and/or stations using highly directional arrays.

A New Method For Broadbanding Antennas

The Flatterer™ is a new, all-electronic, method for broadbanding antennas. Unlike conventional methods of broadbanding, the Flatterer does not require the use of large coils and capacitors.

The main advantages of this patented approach are that the overall efficiency is significantly higher than that of conventional methods and, for medium to high powered stations, the cost and size of components is materially reduced.

The system can treat difficult antenna problems that have, heretofore, required the use of sophisticated multi-section networks having size, cost, and efficiency factors that have made their use impractical. Furthermore, the Flatterer system may be used to provide the desired performance at the far field; i.e., at the station's main target, not merely at the common point.

This new technique has been in service on a full time basis for some six months at one major 50 kw station.

Asymmetry, the Problem of Broadbanding AM Transmitting Antennas

The effect on a narrowband antenna on an AM wave was known by the earliest pioneers of AM technology as reported in a classic paper, "Operation of AM Broadcast Transmitters into Sharply Tuned Antenna Systems", Wm. Doherty, in the Proceeding of Radio Engineers, July, 1949, pp. 729-734.

This paper demonstrates the fact that a narrowband antenna disrupts the symmetrical transmission characteristic of the upper and lower sidebands causing two problems:

1) Loss of modulation and,

2) Envelope distortion.

On the other hand, if the response to the two sidebands was symmetrical, only the loss of modulation would be experienced and simple preemphasis of the higher frequency audio components, fed to the transmitter, could be used to cure the problem. However, antenna response is not symmetrical and attempts at compensating for loss of modulation by preemphasis would greatly increase distortion. Thus, the most serious problem in broadbanding antennas has been asymmetry and correction circuits always have, hertofore, been required to handle the station's full output power.

Figure 1 shows, in simplified block form, the basic structure of the conventional method for broadbanding an antenna. It is obvious by just glancing at the Figure 2, that the older (Figure 1) method is conceptionally simpler and there is, apparently, much less equipment involved. Actually, for a low powered station, without an unusual antenna problem, the conventional antenna broadbanding approach would be superior to the new method in terms of cost and simplicity.

However, for high powered stations, the broadbanding network of Figure 1 can be expensive and provide less than ideal performance. This is especially true where directional arrays are involved, or the station operates on a low carrier frequency.

It is significant that the power wasted in the high powered broadbanding network is at the output of the transmitter and therefore is net of the inefficiencies in the transmitter. Furthermore, since these coils and capacitors are necessarily large in size, adjustments are difficult to implement. It is obvious that the complexity of such networks, in order to maintain reasonable size and efficiency, must be extremely limited, thus, in turn, restricting the complexity of the antenna problem that can be effectively treated.

Flatterer Block Diagram

Figure 2 shows a block of the overall "Flatterer" system.

A low powered signal, at an IF frequency of either 1.4 mHz or 1.6 mHz, is produced by the miniature transmitter. The choice of frequency is dictated by the carrier frequency of the station so that if the station's carrier frequency is close to the 1.4 mHz or a subharmonic thereof (700 kHz), then the 1.6 mHz IF is selected.

This low powered wave may be a monophonic wave or either type of AM Stereo signal now in the marketplace. This low powered wave should be fully processed and incorporate the desired preemphasis characteristic.

The low powered (approximately 1 mw) modulated wave is then fed to a six section network. The purpose of this broadbanding network is to generate a signal, that, when passed through the antenna, will have the same spectrum characteristic as the wave generated by the miniature transmitter. Thus, the wave at the output of the broadbanding network should have a mirror symmetry frequency domain characteristic with the carrier frequency being the reference, center frequency.

For a perfect monophonic wave, or an AM Stereo wave where L=R, the sidebands at the output of the antenna should be exactly equal. However, due to the lack of bandwidth of the antenna one of the sidebands is attenuated relative to the other sideband and also altered in phase producing distortion and loss of modulation as above discussed.

If, for example, a 10 kHz tone produces, at the output of the antenna, a lower sideband that is 5 db weaker than the upper sideband, the action of the Flatterer system should be to raise the level of the lower sideband, relative to the upper sideband, by 5 db. The means for doing this is to utilize the broadbanding network to produce a wave with mirror symmetry relative to the antenna's characteristic, which is then power amplified by the transmitter to produce a high powered wave that, when passed through the antenna system negates the asymmetry produced by the antenna.

Each of the six sections of the broadbanding network comprises a highly stable tuned circuit where the exact center frequency can be adjusted ± 15 kHz from the 1.4 mHz IF. The Q can be adjusted over a range of approximately 3 to 60.

Each tuned circuit section has a 50 ohm input and output impedance using isolating emitter follower circuits to insure freedom from interaction between the sections. Since the Q and tuning adjustments cause the output level to vary, an AVC circuit is provided which can be switched in when the network is being adjusted so as to maintain a constant output level.

The output wave of the broadbanding network can be adjusted for precise mirror symmetry of the antenna's characteristic. In other words, the output of the broadbanding network is the same characteristic of the antenna but with a mirror symmetrical characteristic relative to the reference frequency; i.e., the carrier.

It should be stressed that the wave at the output of "Broadbanding Network" is a "hybrid" modulated wave (i.e., a wave having both an envelope modulation component and an angular (FM or PM) component) even when a pure AM wave is fed to it. Thus, even when perfectly symmetrical dsb waves are produced by the miniature transmitter, the output of the "Broadbanding" network will have a PM component as well as the envelope modulation component.

Accordingly, no matter what the type of modulated wave being transmitted, the net result will be that the output of the broadbanding network has both envelope and phase modulated components and, therefore, it is necessary that any subsequent power amplifier be able to handle both components of such hybrid modulated waves.

A high efficiency system (Kahn U. S. Patent 2,666,133) has been developed for high efficiency power amplifier of hybrid modulated waves called the Envelope Elimination and Restoration system (EER). The Flatterer uses such technology so as to make the system compatible with transmitters using Class-C amplifiers.

Of course, the Flatterer wave, as it exists at the output of the broadbanding network, could also be power amplified with a linear power amplifier. But linear amplifiers are not commonly used in broadcasting because they are far less efficient than non-linear amplifiers, such as, Class-C amplifiers.

The output of the broadbanding network is phase demodulated and the resulting audio wave is equalized and time delayed in order to be compatible with the associated transmitter. This audio wave is then used to phase modulate a carrier wave operating at the carrier frequency of the station.

This phase modulated wave is then power amplified to approximately a 4 watt level so that it can be used to excite the transmitter.

The output of the "Broadbanding" network is also fed to an envelope demodulator which produces a second audio wave. This wave is time delayed in a fairly wideband time delay network so as to insure wave shape integrity. Generally, the time delay in this path is unnecessary but certain transmitters do require more time delay in the modulator path.

Method of Adjustment

If the antenna being broadbanded is a non directional antenna, an RF sample from the common point is, in most cases, satisfactory for monitoring the action of the Flatterer. However, for directional antennas, a sample of the signal in the main lobe, preferably a mile or two away from the antenna, is necessary for accurate adjustment. In any case, adjustments are fairly simple and can be done in a short period of time.

The unit should be tested with a single tone having a frequency equal to at least the highest frequency being transmitted. (If a stereo transmitter is involved the L-R channel should be disabled, as symmetrical sidebands are required for the following adjustments.) Actually, it is desirable to start at a frequency somewhat higher than actually found in the program wave, so as to insure that a proper "roll over" phase characteristic results.

Say a tone having a frequency of 15 kHz is used, and say, for example, the lower sideband shows 5 db less amplitude than the upper sideband, the first section of the six section broadbanding network should be tuned to 15 kHz below the 1.4 mHz IF. This centers this tuned circuit on the 15 kHz lower sideband. The circuit should be adjusted for minimum Q and then the Q should be raised until the lower sideband, as viewed on a spectrum analyzer, just equals that of the upper sideband.

The frequency of the tone is then reduced to say 12 kHz. Assume that the 12 kHz lower sideband is 1 db below that of the upper sideband, the second tuned circuit should then be adjusted to approximately 12 kHz below the carrier and the Q adjusted so that the 12 kHz lower sideband just equals the amplitude of the upper sideband.

The test oscillator is then reduced in frequency to 10 kHz and the third tuned circuit is adjusted for equal sidebands. Generally, only two or three of the tuned circuits need be adjusted out of the six provided, in order to provide flat overall response.

The entire frequency range is then swept to make certain that the spectrum is flat, say within ± .2 db.

It should be noted that if the spectrum is not symmetrical for low frequency signals; i.e., below 1 kHz, the antenna is, in all probability, not the problem because this would take Q's in excess of values expected from such antennas. In such situations the sideband asymmetry is probably caused by poor neutralization of the transmitter, producing incidental phase modulation. This incidental PM component combines with the envelope modulation resulting in the asymmetrical sideband characteristic.

FIG. 1

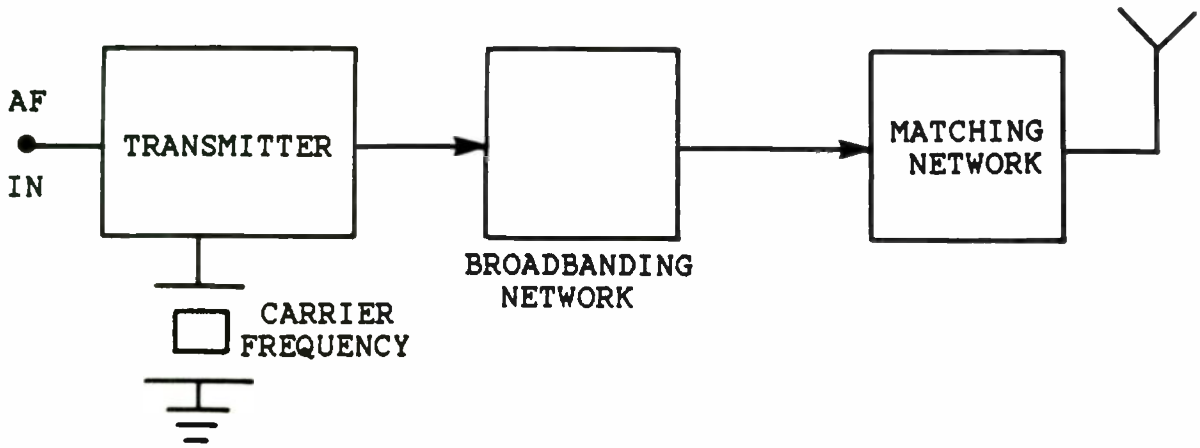
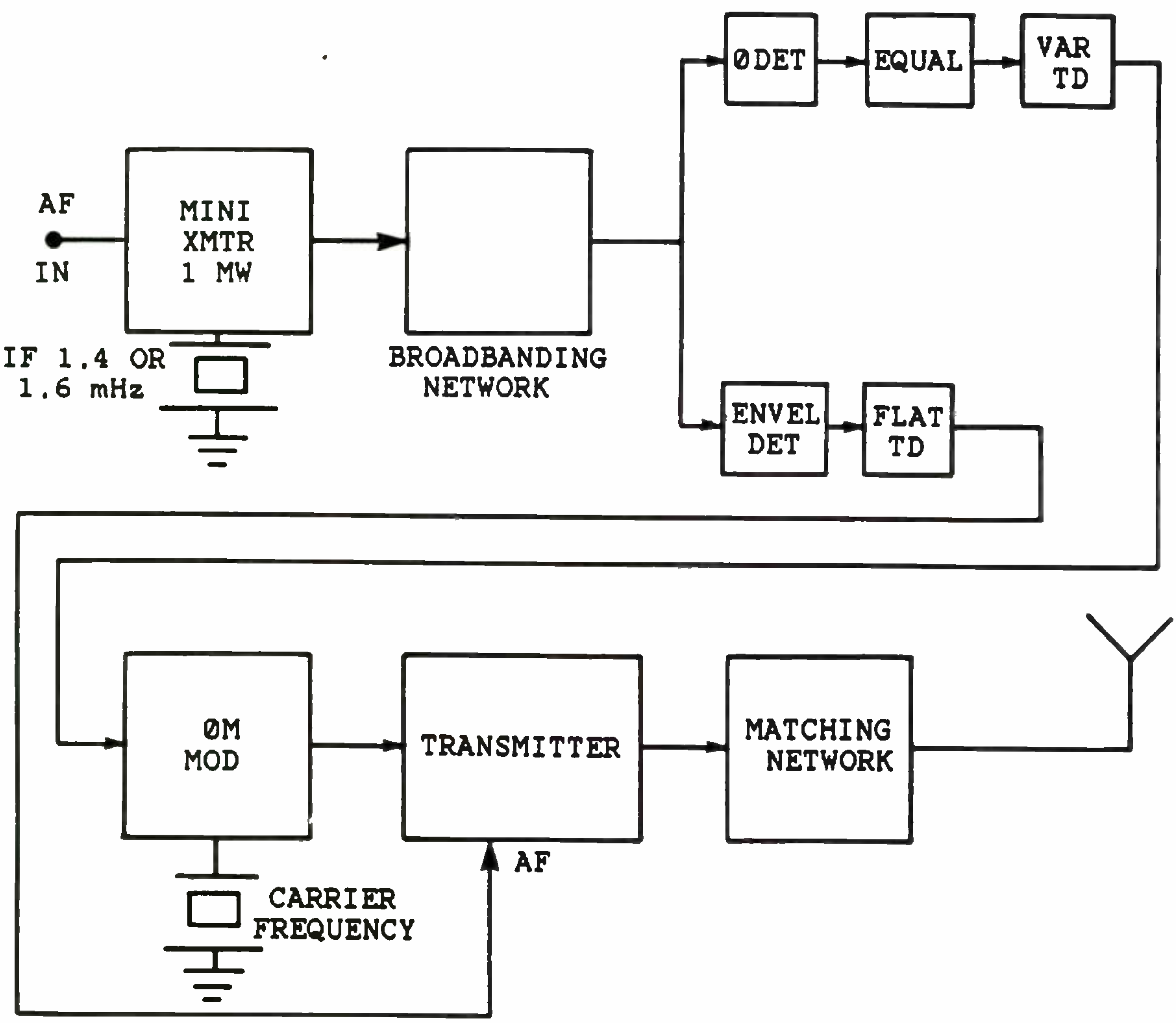

AF
IN
TRANSMITTER
CARRIER
FREQUENCY
BROADBANDING
NETWORK
MATCHING
NETWORK

FIG. 2


ØDET
EQUAL
VAR
TD
AF
IN
MINI
XMTR
1 MW
IF 1.4 OR
1.6 mHz
BROADBANDING
NETWORK
ENVEL
DET
FLAT
TD
ØM
MOD
TRANSMITTER
MATCHING
NETWORK
AF
CARRIER
FREQUENCY

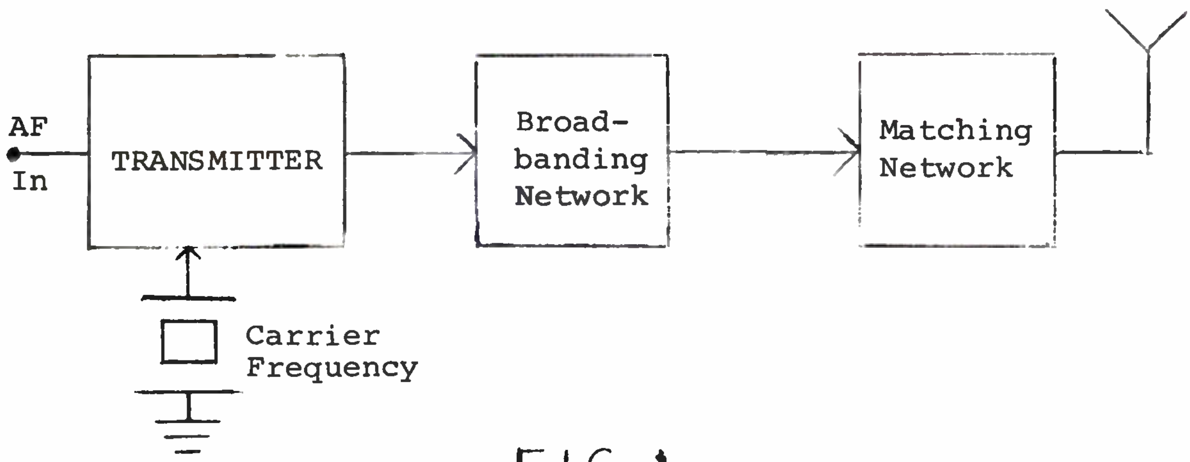

AF
In
TRANSMITTER
Broad-
banding
Network
Matching
Network
Carrier
Frequency

FIG.1

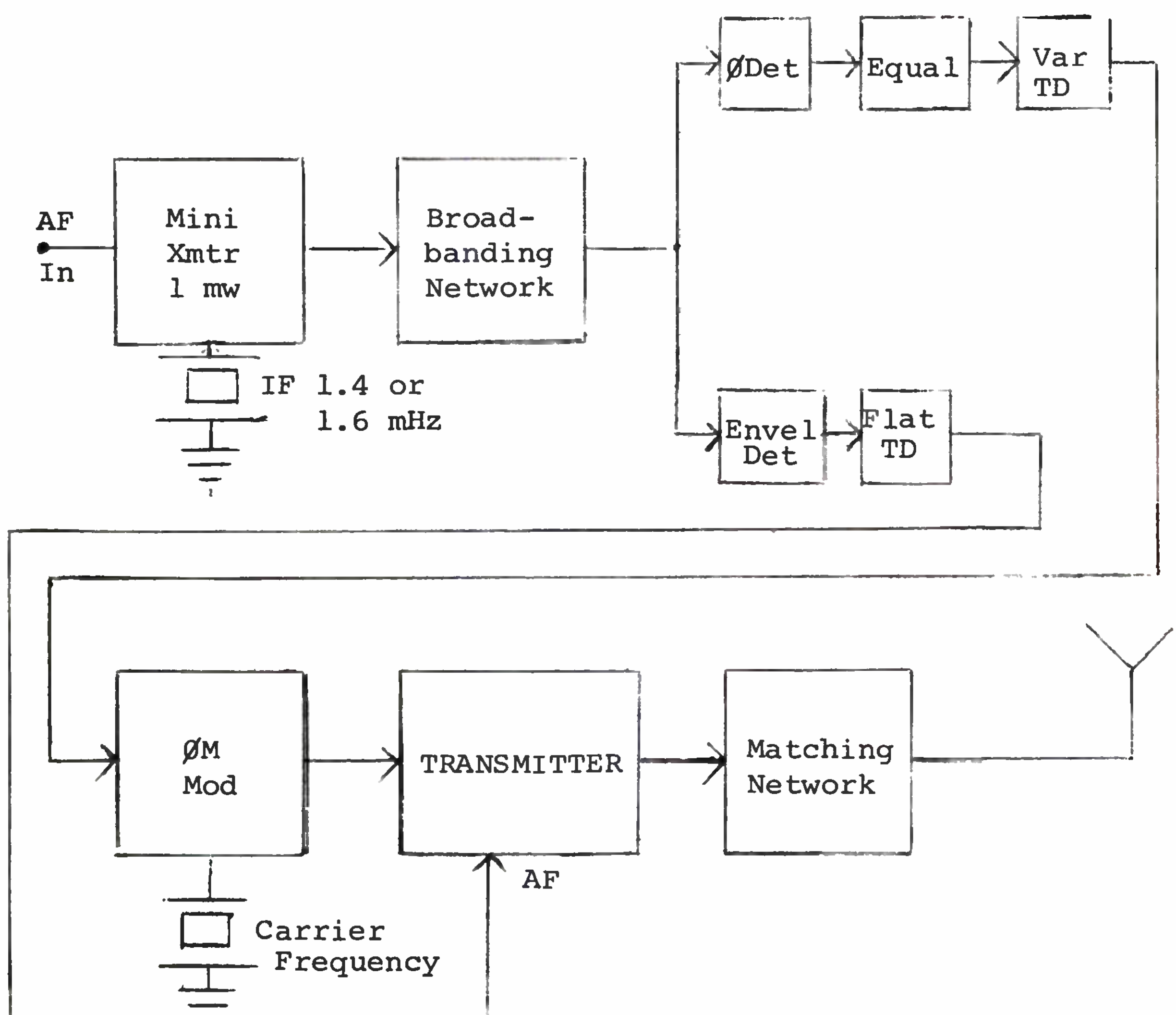

AF
In
Mini
Xmtr
1 mw
Broad-
banding
Network
IF 1.4 or
1.6 mHz
ØDet
Equal
Var
TD
Envel
Det
Flat
TD
ØM
Mod
TRANSMITTER
Matching
Network
AF
Carrier
Frequency

FIG.2

DIAL TELEPHONE TRANSMITTER REMOTE CONTROL

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Two major points will be addressed in this paper in relation to transmitter remote control using the dial telephone system as the interconnecting circuit. These points are -

- ☐ FCC Questions, Is It Legal
- ☐ How to do it

The dial telephone system presents a unique and versatile means of operating a broadcast transmitter. Until the adoption of FCC Docket 84-110 on November 19, 1984 such use of the dial telephone system for remote control of a transmitter was not possible under FCC Rules and Regulations. This Docket and the Rules that were changed will be discussed. The current Rules are reproduced below, and finally a discussion of possible hardware to actually accomplish the task will be presented.

Cost can be a major reason for selecting the dial telephone system rather than dedicated telephone lines for remote control. Dedicated lines have become, in many areas of the country, extremely expensive. Last year a quotation was obtained for an unconditioned data circuit from our offices in San Jose, CA to the nearest FM transmitter site, a distance of 5 or 6 miles. Such a circuit would have passed through two central offices and had a monthly cost of approximately $135.00, plus the installation cost. For this station's transmitter site, the dial telephone was already installed and was used for voice communications. As a result, to use the dial telephone line for transmitter remote control would not have noticeably increased station operating costs.

Another major advantage of the dial system is the ease of access. You simply make a telephone call. The studio or control point equipment can be as simple as the Touch-Tone [r] telephone on the desk in the studio.

☐ FCC Questions, Is It Legal

The FCC, through Docket 84-110, removed all previously existing remote control rules and replaced them with section 73.1400 and 73.1410. These Rules apply to all classes of broadcast stations, AM, FM and television.

For reference purposes, the following is a reproduction of these two sections -

---

73.1400 Remote control authorizations.

(a) An AM, FM, or TV station transmission system may be operated by remote control using the procedures described in 73.1410.

(b) No authorization from the FCC is required to operate the transmission system of an AM station operating with a nondirectional antenna, FM station, or TV station by remote control. Authority to operate an AM station using a directional antenna system by remote control is obtained using the following procedures:

(1) An application for a construction permit to erect a new directional antenna or make modifications in an existing directional antenna, subject to the sampling system requirements of 73.68, may request remote control authorization on the permit application FCC Form 301 (FCC Form 340 for noncommercial educational stations).

(2) A licensee or permittee having a sampling system in compliance with the provisions of 73.68(a) must request remote control authorization on FCC Form 301-A, and submit information showing that the directional antenna sampling system has been constructed according to the specifications of 73.68(a).

(3) A licensee or permittee of a station not having an approved directional sampling system in compliance with the provisions of 73.68(a) must request remote control authorization on FCC Form 301-A, and submit information showing that the directional antenna is in proper adjustment and further showing the stability of the antenna system during the 1-year period specified in Section II of Form 301-A.

(c) Whenever a remote control point is established at a location other than at the main studio or transmitter, notification of that remote location must be sent to the FCC in Washington, D.C, within 3 days of initial use of that point. This notification is not required if responsible station personnel may be contacted at the transmitter or studio site during hours of operation when the remote control operator is elsewhere.

73.1410 Remote control operation.

(a) Broadcast stations operated by remote control must provide at remote control points sufficient control and operating parameter monitoring capability to allow technical operation in compliance with the Rules applicable to that station and the terms of the station authorization. AM stations that are required to change modes of operation during the broadcast day must provide sufficient redundancy to assure that such mode changes actually occur.

(b) The remote control system must be designed, installed, and protected so that the transmitter can be activated or controlled only by licensed transmitter operators authorized by the licensee.

(c) The remote control and monitoring equipment must be calibrated and tested as often as necessary to ensure proper operation.

(d) The remote control system must be designed so that malfunctions in the circuits between the control point and transmitter will not cause the transmitter to be inadvertently activated or to change operating modes or output power.

(e) Whenever a malfunction causes loss of accurate indications of the transmitter operating parameters, use of remote control must be discontinued within 3 hours after the malfunction is first detected. If the station is found to be operating beyond the terms of the station authorization and such malfunction cannot be corrected by remote control, station operation must be immediately terminated.

(f) AM stations may use amplitude or phase modulation of the carrier wave for remote control telemetry and alarm purposes. FM stations may use aural subcarriers and TV stations may use either aural subcarriers or signals within the vertical blanking interval for telemetry and alarm purposes. Use of such remote control signals must be in accordance with the technical standards for the particular class of station.

The first thing to be noted in these Rules is the lack of a fail-safe requirement. In the Report and Order for this Docket, the Commission stated that nearly all the respondents expressed that the primary difficulty with remote control operation was the fail-safe requirement. Thus, the requirement for fail-safe circuitry was removed and is not included in the new Rules. In the Docket, the Commission took note that the Communications Act of 1934 requires that a licensed operator have supervisory control over the transmitter during all periods of operation. The operator should be able to terminate operation if the station is the source of harmful interference, or is operating inconsistently with law or treaty. Note that there is no "how to" methodology in these Rules.

An area of possible confusion is that of notification of remote control operation. No formal filing is required except for AM directional stations. Only notification of operation is needed. If the remote control point is the studio, no notification is required. A directional AM station needs to file Form 301A for remote control operation [see 73.68(a)]. The Commission does want to have someone to contact if there is a problem. While the terminology can be read to imply that "if responsible station personnel may be contacted at the transmitter or studio site during hours of operation", then notification is not required. I do not believe this is meant to imply that a remote control point can be established anywhere, without notifying, so long as someone is at the studio or transmitter. The Communications Act of 1934 does require that an operator exist.

The new Rules no longer advise what parameters to observe. They only specify that sufficient control and operating parameter monitoring capability be provided to insure compliance as needed at the remote control point. Gone are the requirements of a meter of equal accuracy as on the transmitter. Meters are not even needed, only monitoring capability. This becomes an important point when we consider how to accomplish dial telephone operation.

The remote control system is to be designed, installed and protected such that the transmitter can be activated or controlled only by licensed transmitter operators authorized by the licensee. Furthermore, the system need only be calibrated and tested as often as necessary to insure proper operation. It is to be designed so that malfunctions in the circuits between the control point and the transmitter site do not cause inadvertent activision of the transmitter or the changing of operating modes or powers. Again, this is not meant to imply a fail-safe. Whenever a malfunction causes loss of accurate indications of transmitter operating parameters, remote control operation must be discontinued within three hours after the malfunction is first detected. If the station is found to be operating beyond the terms of the station authorization and such malfunction cannot be corrected by remote control, station operation must be immediately terminated.

While the Rules do not provide specific guidance for this type of operation, as had been the case in the past, efforts have to be made to insure that the intent of the rules are met. The Communications Act requires that a station have the ability to terminate operation. This is considered the ability of having absolute carrier control. While not a fail-safe, as existed in the past, it is to have the ability to cease radiation at will. From discussions with members of the Commission's staff and from presentations made elsewhere by members of the FCC staff, absolute carrier control can be accomplished in a number of ways. The dial telephone system, however, cannot be construed as given absolute carrier control. This is based upon the assumption that use of the dial telephone system is only of an intermittent nature. The dial telephone circuit would only be used when it was desired to observe transmitter operating parameters or when the control system wished to report a problem by initiating a telephone call.

In practice it can be said that, if programming is not present at the transmitter site, there is no need to have that transmitter on the air. Thus, absolute carrier control can be established by the presense or the lack of program material to the transmitter. In the simplest terms, the carrier-operated squelch relay of an aural STL receiver, or the simple presence of program audio, will provide an acceptable means of providing absolute carrier control. One member of the FCC staff has even used the example of "a carefully aimed cannon" as a way to terminate operation. The concept is simply to have a way to terminate radiation.

The new Rules also do not tell what form of monitoring is to used, not even that meters are needed or required. General guidelines from this proceeding for broadcast transmitter operation by remote control were that the licensee be assured that -

A. An operator is on duty

B. The transmitting system operates properly

C. The Commission can contact station personnel during hours of operation

Two additional areas need to be mentioned. These are the EBS requirements and tower lights monitoring.

All stations are required to be at least a "non participating" station in EBS. This means monitoring, airing an appropriate message, and terminating operation when a national alert exists. If the station is a participating station, it would then air the national alert.

Tower lights must be observed. This, of course, can be a visual observation, and may not involve the remote control system.

These two points are included so that they are not overlooked in considering remote control operation.

Given all of the above, a careful assessment must be made of "how to" use the dial telephone system.

☐ How to do it

The dial (or switched) telephone system has two basic functions. It provides -

Voice Communications

Signaling

Signaling is dialing, the ringing of a telephone, or simply the addressing on a selective basis of one location. To use this system for transmitter remote control, or for any remote control application, it must take advantage of these two basic characteristics.

With today's technology, the use of digital techniques has become the simplest and most cost effective method of accomplishing many tasks. A microprocessor or microprocessor-based technique is the best way to accomplish dial remote control operation. Figure 1 is a block diagram of such a transmitter remote control unit.

At first glance this block diagram looks much like a personal computer. It contains a central processing unit (CPU), memory, clock and a variety of interfaces. The interfaces, the software that operates the unit, and the physical packaging are what makes the resulting unit different from a personal computer. These are what makes it a transmitter remote control unit.

Data acquisition, the gathering of information, for use on the dial telephone system, has a direct effect upon the software used in this application. With a dedicated interconnecting circuit operation, whether it is telephone or subcarrier, the method of handling information has typically been direct transmission of that information. With the dial interconnect, storage of information becomes critical. Information will be reported at a later time than is the case with dedicated interconnect. With the dial interconnect, the connection must first be made before it can be transmitted. The time required to make the telephone connection is not unknown, and can be from seconds, to minutes or even hours. With a dedicated line system, the transmission time is known and is set by the system design. As a result, the various methods of handling data must be tailored to dial operation.

For a unit to function, it must have the ability to observe metering (analog) information, status (go/no-go or digit) information, and have command capability. To observe operation of a transmitter, it is simplest to meter power output. After all, this is the end product, a signal on the air. While go/no-go indications can give a "yes/no" indication, it lacks the absoluteness of an actual power reading. It is better to know that you are at 86% of full power, rather than the station is simply still on the air. Knowing that you are within 1% of a lower parameter can result in your taking action to correct the problem before a decision must be made to go off the air. Note that the Rules no longer require having the ability to actually adjust power output, only that you stay within the range existing of the class of license. It is best to maintain the ability to adjust power output so that the maximum power output is in use at all times.

In the design of the VRC-1000 it was elected to take full advantage of the characteristics of the dial telephone system. The use of Touch-Tone [r] or DTMF signaling and the use of a synthesized voice output were selected as these are the two things the dial system best transmits. More importantly, this would enable a simple telephone or DTMF pad combined with a pulse-dial telephone to be the control end of a system.

It was determined that the maximum flexibility in speech output should exist at the most affordable cost level. To accomplish this, a specific LSI talking device was selected. This device uses minimum digital storage per word while maintaining recognizability and permitting direct addressing of voice generation output. It permits word groups, or sentences, to be addressed at CPU speeds, while requiring only one 27128 memory device per 100 words of capability. The VRC-1000 was created with fixed framework outputs that can be set up to output the words needed for each metering, status, and command channel. A typical framework for a metering channel is two words to identify the channel and two words for the units of measure. The unit itself generates the numeric value as calibrated by the user. Even with the numeric value, attention must be paid to how it is generated. A verbal output of 1140 ("one, one, four, zero") does not communicate as easily as an output of "One Thousand One Hundred Forty". For a metering channel, a possible output can then be "Metering Channel Three, Power Output, One Thousand One Hundred Forty Watts".

Other considerations relate to how the telephone system or telephone instruments behave. As an example, when a command is issued, a verbal feedback verifying that the command has been received by the unit is desirable. This is equivalent to the raise or lower button of a conventional remote control system being illuminated. Some telephones, however, do not permit you to hear inbound audio when DTMF is being sent, thus eliminating any direct feedback. In doing power control, it could be impossible to actually monitor a metering channel and issue a command at the same time, as the inbound audio could be blocked by the DTMF being sent to cause the command. For this reason, in the VRC-1000 momentary commands were restricted to a user defined length of time, not a continuous output of the duration that a command is sent.

There are a number of things in the speech domain that had to be considered. A variety of operations relating to the handling of the telephone system require attention. As examples, the VRC-1000 will dial up to five telephone numbers to report a problem with metering or status channels. To automatically dial, it is preferable to have the unit recognize dial-tone before dialing. This permits the telephone line at the site to be used for normal service when someone is there as well as by the unit. If dial-tone is not recognized, the unit would simply dial anytime it was activated. By incorporating dial-tone recognition in the unit, this problem is eliminated. In some parts of the country the local telephone company is using equipment of sufficient age that it does not issue precision or correct dial-tone signaling. Of course, the human ear can be very forgiving in determining that dial-tone is present. A device such as the VRC-1000 must be sure, and requires a standard dial-tone. To handle situations where non-precision dial-tone is used, the VRC-1000 has incorporated a software switch that turns on or off dial-tone recognition. Likewise, to function in most parts of the country, the unit must be able to itself outward dial using either DTMF or pulse dialing.

Voice has the attributes of being easy to transmit and easy to receive. All that is needed is a telephone to listen to the unit. This makes access possible from any telephone. This type of access is also inexpensive. It, however, does not make for the fastest mode of communications. Speech must be presented at a given rate. That rate is much slower than most digital transmission schemes. With the unit using a microprocessor, having both voice and data outputs is possible. Note that a modem is shown as an option in Figure 1. With a data output, it is possible to display information on a terminal or personal computer. This opens the possiblities of automatic logging in addition to the observation of all information at one time on one presentation.

Having a microprocessor, the possiblity of automatic commands becomes immediately apparent. The VRC-1000, as currently configured, permits multiple level limit checking on each metering channel, and observation of both states of all status channels. These points can be used to activate commands automatically. Time of day can also be used for the same function.

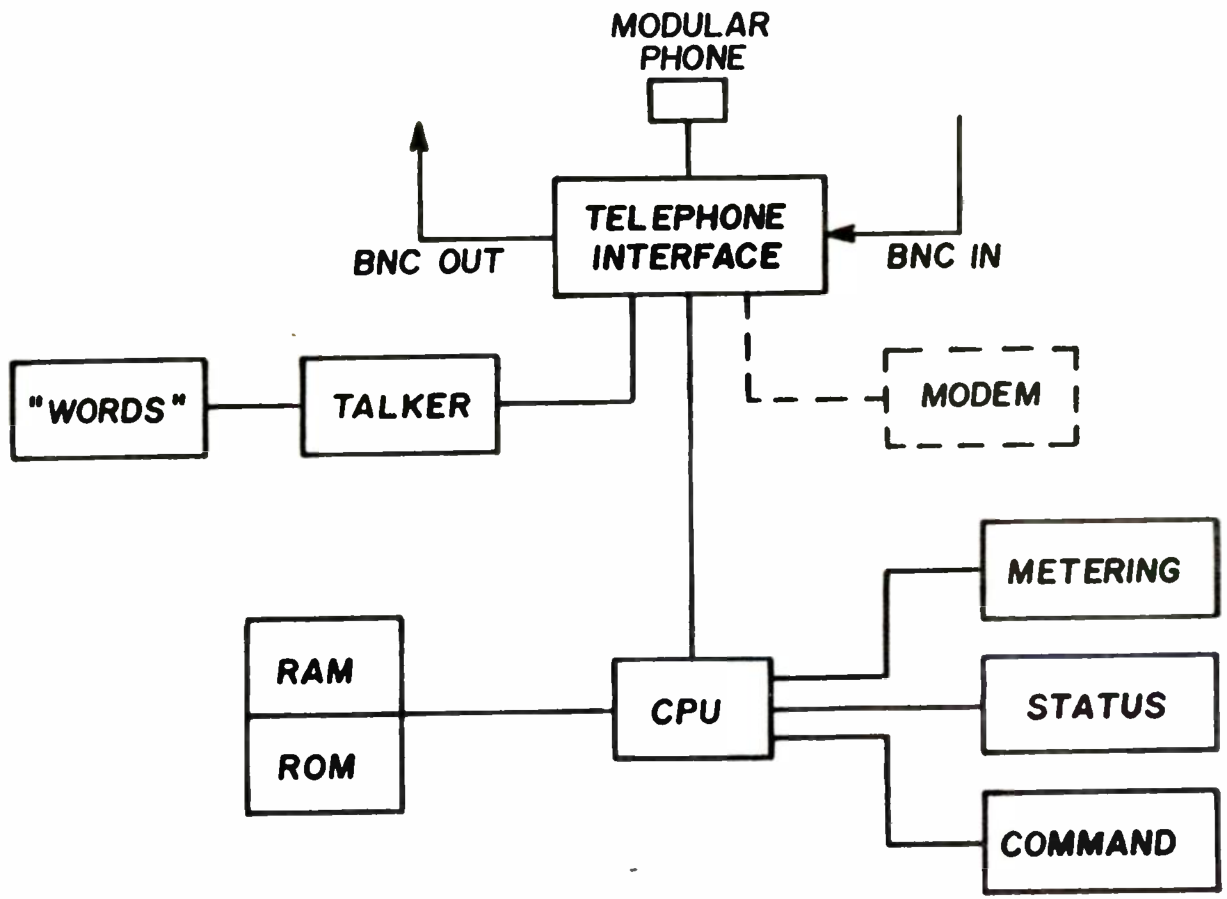
As a manufacturer, we determined that a microprocessor-based product of this type requires attention to not only the current needs of users but also the future needs. To this end, the VRC-1000 was designed such that the software, or firmware, could be easily changed or updated in the field by the user through the simple exchange of a single memory device.

Not shown shown in Figure 1 is an internal microphone. Once the telephone connection is made, it makes sense to include the ability to provide an audible means of monitoring the area near the unit. If nothing else, it enables listening to the blower noise from the transmitter - a good sign, you are still on the air.

Among the power sources is a lithium battery. This was included, and is hardwired into the unit, to maintain both the clock and solid-state memory with an AC power loss. The basic type of devices used throughout the unit were selected to permit the lowest possible power consumption. Not only does this limit the amount of heat generated by the unit, but also permits a lengthy memory retention without external power. An 18-year shelf life mil-qualified lithium battery was selected. It will maintain memory and the clock for a minimum of two years without AC power being applied. This also means that the state of all commands is maintained for the same length of time, as well as all alarms that may be in memory. For actual operation, a separate battery supply exists to power the complete unit for an extended period.

The physical packaging of a unit of this type is as important as other areas. The VRC-1000 is contained in a single steel chassis. This provides excellent magnetic and RF protection. All metering and status inputs enter via four-section filters.

At the time of presentation of this paper, it is planned that an actual demonstration will be given of a VRC-1000. A unit will be activated locally or a call will be made to a unit somewhere in the world. The unit to be called will not necessarily be one located in North America, for the concept has been proven to work successfully on an international basis.



VRC-1000 FUNCTIONAL BLOCK DIAGRAM

FIGURE 1

GENTNER
RF PRODUCTS DIVISION

A NEW ERA IN VIDEO MEASUREMENTS

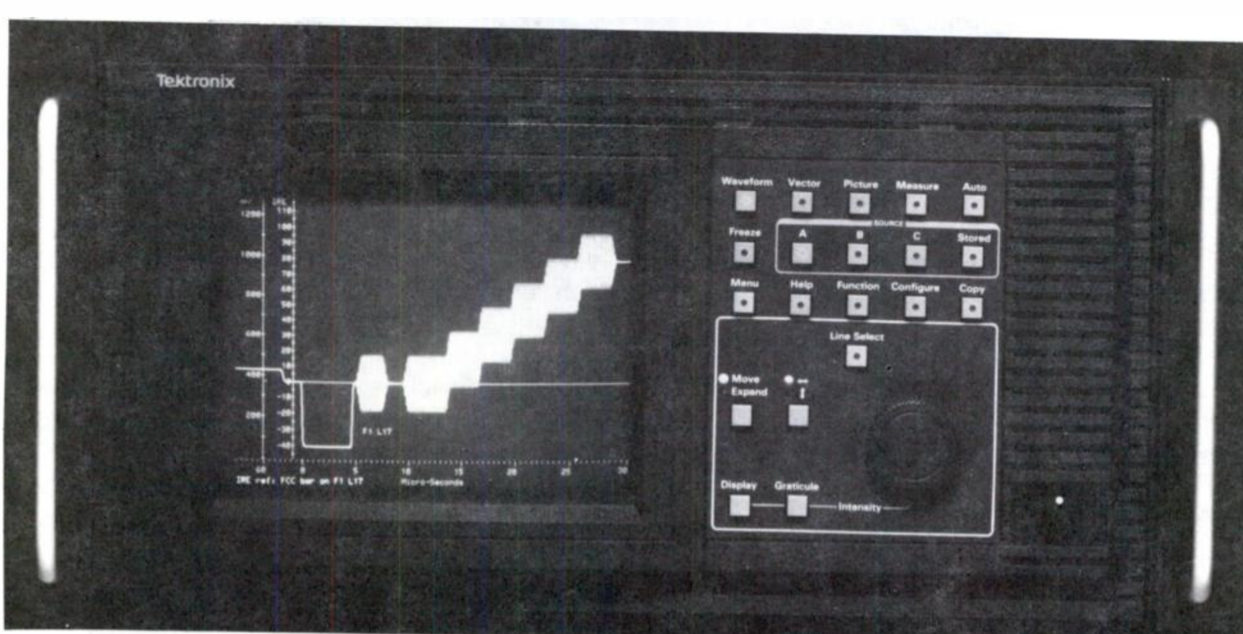
By Larry Harrington

Tektronix, Inc., Beaverton, Oregon

For test and measurement equipment needs, television engineers have previously had to make the choice between manual and automated test equipment. Most manual equipment requires technically competent users and some set-up time. The measurement accuracy is often dependent on the experience of the person making the measurement. Later, even with a photo of the waveform, the user is often unsure if the measurement was made properly.

Automated test and measurement equipment is less dependent on the operator, but usually has been too slow for making adjustments based on measurement values. It has also been bulky and expensive. For these reasons, most broadcasters have continued to use manual video test equipment in all applications.

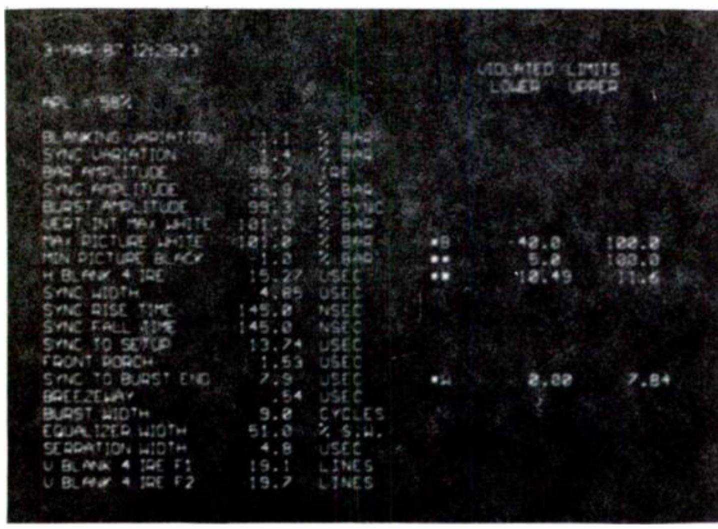
Today, the broadcaster has a new and better solution for his test and measurement needs — the Tektronix VM700 Video Measurement Set. The VM700 makes measurements quickly and automatically, while providing manual capabilities for further analysis of problems.



The VM700 has been designed as an easy-to-use video measurement instrument with full measurement capability. A front panel with a few buttions, a multi-use knob, and a touch screen CRT make this possible.

This product is difficult to describe in traditional terms because of the extensive capability in a simple package. The design objective was to provide a comprehensive video measurement product which meets the broadcast engineer's needs in an easy-to-use package at a moderate price. Technology advances in both hardware and software (and a lot of creative engineering) have enabled the accomplishment of these objectives in the VM700.

The VM700 is first of all an automatic measurement set. Most standard video measurements can be made quickly and accurately with the touch of a single button. Front panel set-ups and judgment calls for noisy signals are no longer a concern. The measurements are made the same way every time. The instrument may also be used for automatic monitoring of all, or a selected group of measurements. The user can specify two sets of limits (caution and alarm) for those measurements. The VM700 will then print a message whenever a measured value crosses one of the limits. Reports containing the measured values may also be scheduled for automatic printing at specified times.

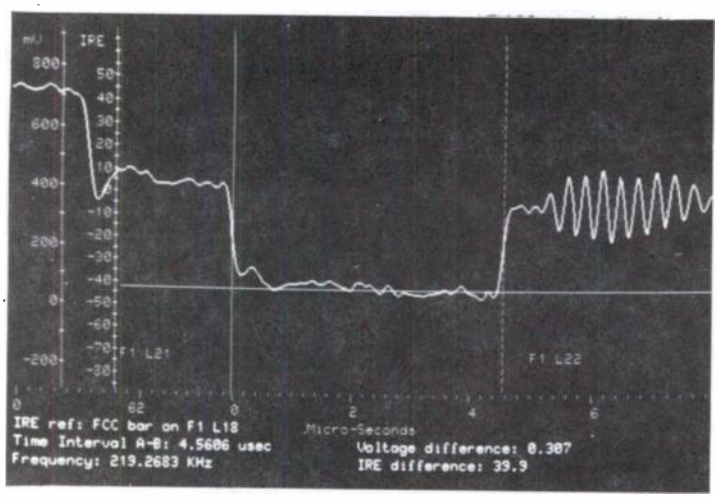


By selecting the AUTO button, the user quickly obtains his measurements. The values of measurements he has previously requested are displayed on the screen. Any values exceeding his specified limits are flagged, and the limit values are also displayed.

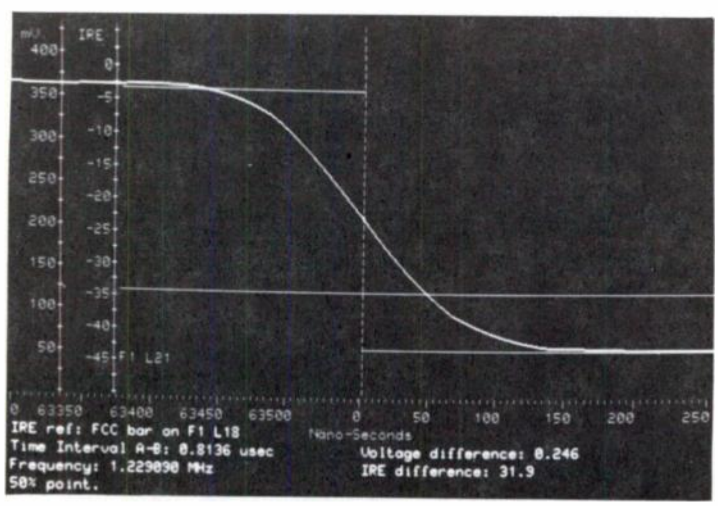
The VM700 can free engineers from routine quality assurance duties and allow them more time to address problem areas, or new facilities. Proof of performance on both permanent and temporary facilities can be done very quickly, and confidently.

For further analysis and troubleshooting, the VM700 is also a digital waveform monitor and a digital vectorscope. It is also a specialized measurement device for measuring group delay, signal-to-noise ratio and other parameters normally requiring separate instruments.

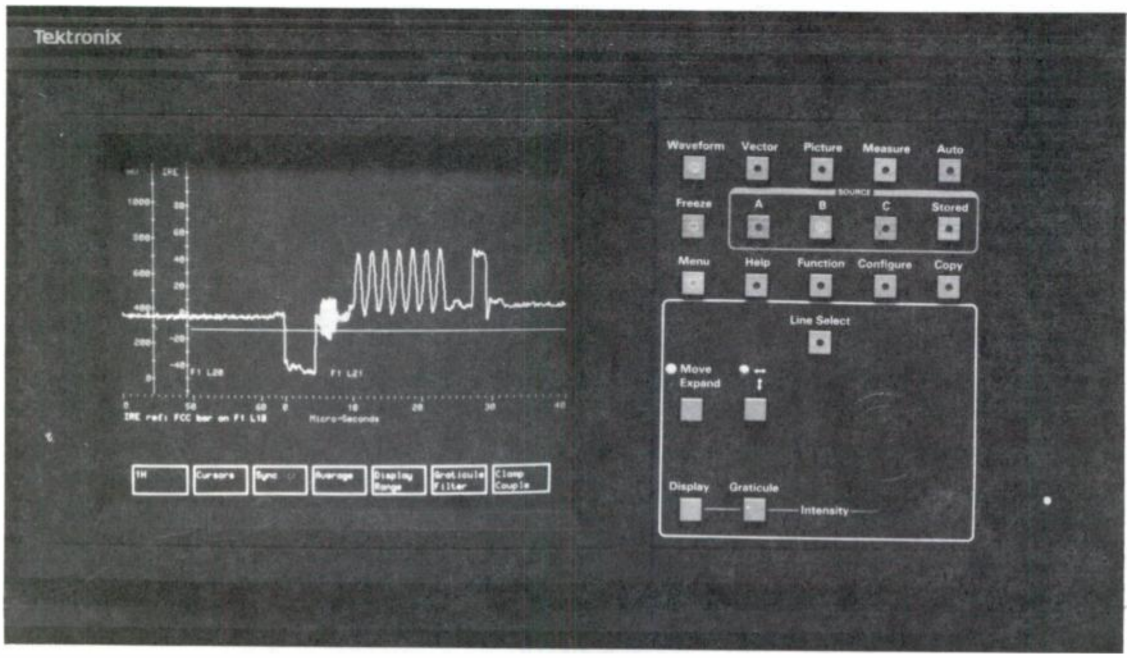
The WAVEFORM button turns the VM700 into a digital waveform monitor. With full touch screen capability and a knob, the user can select and expand any part of a video signal. The electronically created graticules track the waveform as it is expanded or contracted. There is no need to calculate time or amplitude from units per division, and the display is always in calibration. Because the display is digital, the trace is always bright and readable, even with the time scale expanded to display only the leading edge of a single horizontal sync pulse. Cursors are available for manually measuring time and amplitude of waveform characteristics. These cursors can also be used to quickly identify the 10, 50, or 90 percent points on transitions for making manual measurements.



The WAVEFORM button turns the VM700 into a waveform monitor with full expansion capability, cursors, and electronically generated tracking graticules.



The digital waveform display provides a bright waveform regardless of the time interval selected.

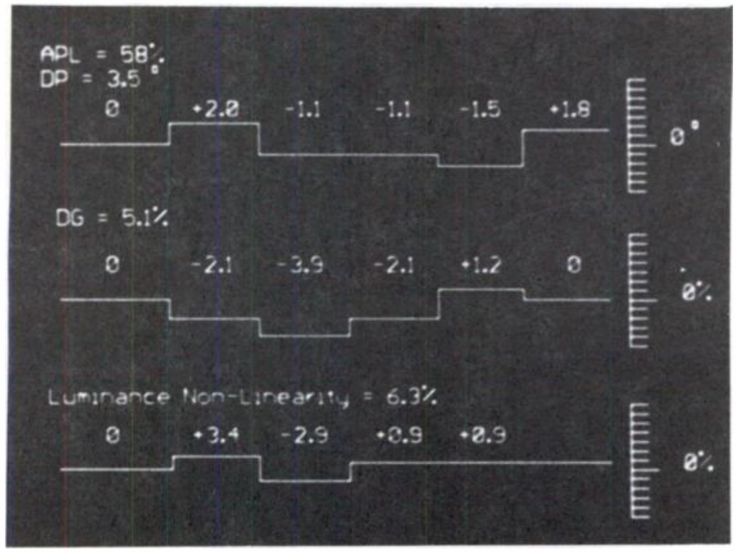


The buttons allow the user to select any of the basic capabilities. The MENU button allows further selection using soft-key selections on the touch screen.

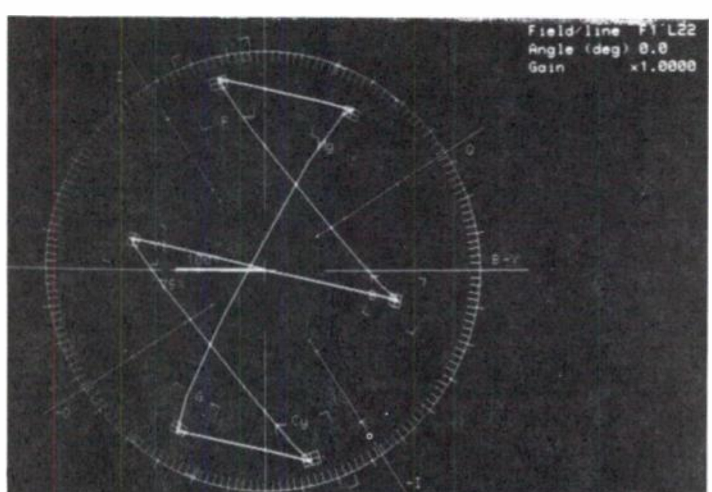
Any line may be selected for display by using the knob for scanning through many lines, or the touch screen for instant selection. The field and line number is displayed immediately following the horizontal sync pulse for each video line.

A picture display is available for verifying that the measurements are being made on the desired signal. This display may also be used for selecting a line of interest from the picture. This can be very useful in evaluating cameras focused on test charts. The VM700 picture display shows a single-field picture, updated at a one-second rate.

The VECTOR button turns the VM700 into a digital vectorscope (i.e., bright, VITS display). With calibrated gain and rotation, the chrominance amplitude and phase can be verified easily. With a soft key on the touch screen, differential phase, differential gain, and luminance nonlinearity can be displayed simultaneously. Thus, adjustments can be made to equipment while observing all three non-linear distortion measurements on only one instrument.

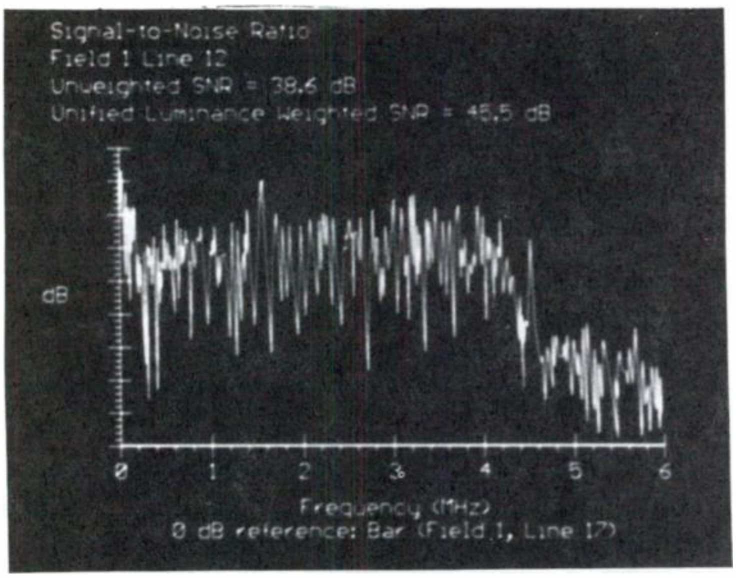


Differential gain, differential phase, and luminance non-linearity are all shown on the same display. Values are shown for each packet or riser as well as peak-to-peak numbers. Thus the user can make adjustments while observing the effect on all three measurements simultaneously.



The VECTOR button turns the VM700 into a vectorscope with calibrated rotation and expansion and some special features such as "color bar search" and DG, DP, luminance on non-linearity measurements.

Noise measurements can be made very quickly. Not only are standard weighted and unweighted signal-to-noise ratio values displayed, but the spectral content of the noise may also be displayed. Now, those assumptions about the spectral shape of the noise (i.e., triangular or flat) can be verified.



The spectral content of the noise is displayed along with the measured signal-to-noise ratio. Assumptions about the noise shape can now be verified.

Group delay can be measured using the SIN X/X or multipulse test signals. This measurement also provides a graph of the delay over the broadcast frequency range, i.e., 0-4.2 MHz.

Most other measurements, including ICPM, can also be made in the measure mode.

Any screen display (including graphs or waveform displays) can be printed with the COPY button on RS-232C compatible printers. For screens with graphic information, Epson and Postscript graphics are supported.

As can be seen from the previous description, there is a lot of capability in one instrument. For those users taking advantage of a range of the capabilities, the task can be greatly simplified using the FUNCTION capability. This allows a user to select and name a sequence of operations under a single touch screen key. This sequence can then be initiated at any time by selecting the named function key. For example, if a technician has a sequence of measurements to be made on the off-air signal, he can select these, as well as automatic hardcopy, and name this sequence "OFFAIR". Now, all he has to do is: 1) select FUNCTION, 2) select OFFAIR on the touch screen, and 3) take the results from the printer.

The VM700 provides the capabilities of many instruments, as well as improved performance, in one package at a reasonable price. Broadcasters now have a new, efficient, and cost effective response to their manual and automatic measurement needs.

Mobile Mast Safety Considerations

By Richard Wolf

President / GM, Wolf Coach Inc,
Auburn, Mass.

A little over a dozen years ago we were approached by the chief engineer of one of the Boston TV stations. They had heard of the work we'd been doing in special applications vehicles, most notable a mast system we had developed for the FAA as part of the Microwave Landing System development program. They were looking into using a 2 Ghz transmitter in the field to send back live news stories, a new and different application of microwave technology. They wanted to know if we could design a mast to get a 2 foot dish 14 feet in the air and remotely position it. Our first ENG mast was born.

Within six months, they were in a situation where they had to move the vehicle to get a better angle for the "live" shot. There was no great problem, the vehicle was in a parking lot, and they were only going to the other end of the same lot. Why not just leave the mast up? Unfortunately as they drove past the shack where the parking attendant sat, they took down the power and phone lines to the shack with the mast top. We had our fist mast accident.

It has long been established that news coverage has its hazards. As the circle of news coverage has grown, these hazards have tended to grow. Masts are taller, the areas that are covered are further away, and in the competitive atmosphere that exists today, the pressure to "get the story" pushes technicians and camera people.

The news vehicle operator must, by nature, also be a creative individual. The degree of creativity in some operators may be a subject of debate, but after seeing the number of ways that vehicles can be damaged, I assure you, there are creative minds at work here. In trying to treat the problem of vehicle safety, we must look first at the symptoms, the damage. The potential for the greatest disaster in these vehicles is in the mast, but there are others ways to have problems.

An overloaded or improperly loaded vehicle can be a hazard on the road. Tire blow outs, strained brakes, worn wheel bearings can all make for an accident waiting to happen. We've even had one van roll on its side when it took a turn too quickly. We don't like to count that as a true ENG accident, however, since the vertical center of mass van was throw off by the half cord of fire wood stacked in the back for the reporters home at the time of the accident.

The mast, because of its movement and height, does still draw the most trouble. Over the years there have been several "creative" ways that they have had problems. The less dangerous problems center around hitting obstructions. The lower the obstruction, the harder the hit. We wondered, if you hit the mast in the collapsed position, just where would the impact be absorbed? That question was answered about five years ago when at 2:00 AM a tired operator went through a driveway instead of around the building, hitting an 8 foot obstruction. The entire roof was opened like a tin can. The mast was destroyed, along with the transmitter, the antenna, the aiming mechanism, and the chances for advancement of the operator at that particular station. Fortunately there were no personal injuries.

By far the most common accident is the sudden removal of the aiming mechanism and antenna from the mast top by a tree or gas station canopy. The broken casting at the base of the aiming mechanism relieves the load, and can be fairly easily replaced.

As the mast goes higher, the potential for more serious problems grows. In one case it grew as the vehicle was going down the highway. A mast valve was not solidly in the correct position, and the mast was slowly being pressurized as the vehicle drove away from a story at the airport. Once the mast got high enough, it hit an overpass, and several sections broke off in the roadway. Again, there where no serious injuries.

It is even possible to have the mast separate without a hit. A misguided technician loosened the bolts holding the collars on a mast, hoping to reduce the drag on the mast, and allow it move more smoothly. Unfortunately the bolts were backed out far enough to disengage the collars, and the next time they applied air pressure to the system the mast was launched a section at a time, landing around the back of the vehicle.

By far the most catastrophic form of personal injury comes from the potential of electrocution and electrical fire. Unfortunately it often takes a spectacular accident to bring home just how serious the injury can be. For me, the event happened while I was still in high school. We were driving past a little league field where volunteer soldiers had spent the weekend putting up lights for the kids. The final act of the day was to stand up a flag pole, and six of them were pushing it up into place. They hit the power lines for the lights with the pole. When we drove by soon afterward, the fire department was watering down the smoldering remains of four of the volunteers. It was a sight and a smell that you can't forget.

People have been springing backs, and slipping with cameras since field news began. In the past 18 months, however, we've all had the real potential for serious injury brought home. One man has lost both his legs, several people have been burned in separate accidents, and one life has been lost. What is different in the recent past? It may be just the law of averages catching up, or more probably that we have become too comfortable with the hazards. It hadn't happened so we all began to think we were immune.

The question now is "what can be done to keep this type of injury from happening again"? The answer is not quite as simple as the question; in fact there are several answers. These answers break into two basic areas, vehicle design, and vehicle operation.

All of the vehicle builders have been aware of the potential for disaster from the beginning. We've tried to anticipate the danger, and design systems to minimize it. Our goal has always been to come up with an "idiot proof" system. Unfortunately the combined best efforts of all involved have only yielded "idiot difficult".

We have two conditions that need to be protected for, the act of deployment, and the possibility of movement while the mast is extended. I'd like to address the second case first. There are a few basic approaches to this problem. All are intended to kill the ignition, stopping the engine if the operator tries to move the vehicle.

The first is a pressure sensing switch in the air line to the mast. In principle, as long as there is any pressure in the line, it means the mast must be at least partially extended, and the vehicle ignition is killed.

The second is a magnetic switch on the roof with the magnet at the mast top, so that as soon as the mast moves from the stowed position, we kill the ignition.

A third alternative is a pin that is tied to a cable tied to the mast top. This allows for a minor misalignment of the mast, and still detect vertical mast movement. An addition that can make things more workable is to tie the kill to the park position on the shifter so the engine can still be run, but will be killed when the vehicle is shifted out of park.

As with any system, there is always a desire for an override. Suppose the vehicle has to move ten feet to get out from behind the shadow of a building. It could take up to ten minutes for the mast to come all the way down, move the vehicle, and go all the way back up. With great fear and trepidation an additional switch can be added to momentarily override the ignition kill. The switch should be keyed, on the outside, and be spring loaded. This forces someone to watch the movement, and walk with it to pace the movement.

One risk is that the vehicle could move too fast, and the vehicle braking action would break the mast. The operator holding the override switch not only paces the movement, they have a tremendous incentive to make sure the overhead area is clear. If power lines are hit, the individual holding the switch is the ground strap. This capability must obviously be weighed against the potential hazard.

By our count the accidents that have resulted in the most serious injuries have happened when the mast was being extended. The options for protection here are just as varied. Lighting to see overhead obstructions is a minimum first precautionary step, with lights on the roof, and of late many have been adding supplementary lighting at the mast top.

The location and type of control valving for mast extension offers several alternatives. One school of thought puts the valve at the drivers door, another inside the curb side door. One school calls for the mast valve to be spring loaded, so the operator will be forced to stand there and watch it go up and down. We used to install spring loaded valves, but we found most of the units with these valves seemed to shortly acquire a stick about 14" long, just the right length to prop the valve open.

Manufacturers have also tried electric solenoid operated valves, so the switching could be done remotely. A suggestion that came up recently was to tie this solenoid controlled valving to a remote infra red control to isolate the operator from the vehicle altogether. This, however, does little or nothing for the individuals in the van, and would be a real problem on rainy nights or with dirty sensors. One must balance this kind of complexity with the probability of a system failure, and the build up of a false sense of safety. There is no question that these capabilities can be accomplished, but the cost and complication are not to be underestimated.

There are proximity devices that use ultrasonics and power sensing to detect obstructions. They have been explored with installations at mast top, with a view toward shutting off air flow if an obstruction is detected, stopping the mast from further extension. Again the consideration for false indications from dirty sensors, or failure to detect the single conductor that could kill, must be balanced against the operational gains.

Another suggestion was for some sort of insulating dome so that if power cables were struck, there would be no electrical path. With the variety of antennas that we are confronted with, and the possible number of positions, we haven't succeeded in coming up with a suitable dome. Besides, unless we could also insulate the mast, and all the cables, we would've made no more than an expensive gesture.

All of the systems can work, and can also be worked around. The key to safety lies in the operation of the vehicles. Working with the systems designed in, or in spite of lack of them, the final safety question is up to the operator. There is no "one size fits all" procedural advice. When we've tried to have a single procedure for all operations, that creativity I mentioned earlier came back to haunt us. The one thing I can suggest to all operations is some sort of formal operational checklist. Pilots, the good ones, live by checklists. Flying has often been described as hours of sheer boredom interrupted by moments of stark terror. From what we've seen there are a good many similarities to news coverage. Days can go by with all nicely scheduled events. Then comes the fire conveniently raging 10 minutes before air time. That's when the actions must be smooth and safe. Make up the checklist with the input of all the crew members, then live by it. Work it until every step is second nature. The specifics of story coverage may change, but the procedures should be the result of planning not panic.

I have a few suggestions for the checklist. These aren't in any order of importance, and different operations will call for additions and deletions, but I would ask that they all be given consideration.

1. Don't pull just off the side of the road and set up. Don't turn into a driveway just off the road and set up. The power wires are usually run just off the road. At least two of the more serious accidents we're aware of were "just off the road". In general stay an extended mast length away from under any power lines.

2. Try to find somewhat level terrain. This way the mast will have a more predictable path of extension, straight up. It'll also be easier on the mast going up and down. I recently saw a vehicle with the right wheels on an 8" curb, and the left wheels on the street. By the time the mast was extended it looked like the top was across the street from the van.

3. When you first park, stop and do a slow count to ten. Recite the names of you kids, the full name of your wife, the full name of your ex-wife, the full name of the workmans comp insurance company, anything to force a pause. The driver has probably just spent the last several minutes rushing through traffic. If it's the breaking story of the day there will be a lot of frantic activity and lots of noise. Ten seconds spent slowing down so the next steps are planned ones can get you on the air faster, and with less risk of making an unnecessary or hazardous mistake.

4. The first thing the operator should do when getting out of the vehicle is look up, then look around. Is the area clear for the mast to go up? Is the area relatively secure? Which direction is the shot to the receiver? If we do have to move a few feet can we without having to bring the mast down? An embarrassed operator in New YOrk had to call the station when he arrived at the scene of a water main break

and jumped out to see what was happening. Someone else on the scene then jumped in the running vehicle and drove off without him. If the van operator is also the cameraperson, things can get to be a push, but a few seconds can save minutes.

If all else fails, and the worst is about to happen, the mast is about to hit power lines, yell the loudest warning possible and bail out. Jump out of the van , don't step out. Get away from the van by at least 50 feet. While the desire to save the damage and embarrassment of an accident is certainly strong, fight it. If the mast is within 10 feet of the wires you can see, it may already be ready to hit the ones you didn't see. Obviously it is desirable to avoid the impact, but again the catastrophic injuries in the most recent news stories happened when the operator saw the problem too late, and tried to deal with it.

No vehicle, or story, is worth losing your legs or your life, it's better to take the fame and inevitable verbal abuse of your contemporaries that a fried van will bring. It's easy to say, and the simplicity of the words can trivialize the fact of the matter, but it's true.

In the final analysis, who do we turn to to solve the problem, once and for all? It is up to us, the vehicle manufacturers, to make a vehicle that can be operated with a reasonable degree of safety. It is up to you, the engineering managers, and the operators of these marvels of modern design to keep them safe. The manufacturer's responsibility is to make safe operation a practical possibility, and the operators responsibility is to make safe operation a practical reality.

THE SOURCES OF ATMOSPHERIC ELECTRICITY

By Ron Nott

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Atmospheric electricity has been around at least as long as man, its most visible and dramatic form being lightning. However, there is always electricity in the atmosphere, albeit in more subtle forms. The sources of this electrical energy are always busy replenishing it. The phenomenon of lightning is surrounded by a plethora of superstition and misinformation. An understanding of the processes involved can do much to explain how lightning and static electricity may be reduced or eliminated.

The last several years have seen more research in atmospheric electricity and lightning than any previous time in history. In addition to basic scientific research, there are two specific reasons for much of this study. First, the stealth bomber, though almost invisible to radar, has proven to be vulnerable to lightning. Its exterior is made of composite materials which could be likened to extremely tough plastic. It is now realized that the aluminum skins of conventional aircraft have been very effective in shielding lightning and its effects from the complex electronic equipment in modern airplanes. Communications and navigation equipment and onboard computers must be protected from the catastrophic effects of atmospheric electricity, so much research is taking place to that end. Second, it has been noted that the electromagnetic pulse (EMP) from a lightning strike is very similar to that of a nuclear explosion except for the much diminished magnitude. See Fig. 1. The risetime is very fast, on the order of a few microseconds as the lightning channel is connected. The trailing edge, on the other hand, shows a slow decay with the curve being known as a reciprocal double exponential. This is due to the resistance of the ionized channel and the depleting energy from the cloud. This is a theoretical curve of an EMP and in the real world is modified by many factors such as the inductance of the channel, multiple paths, etc.

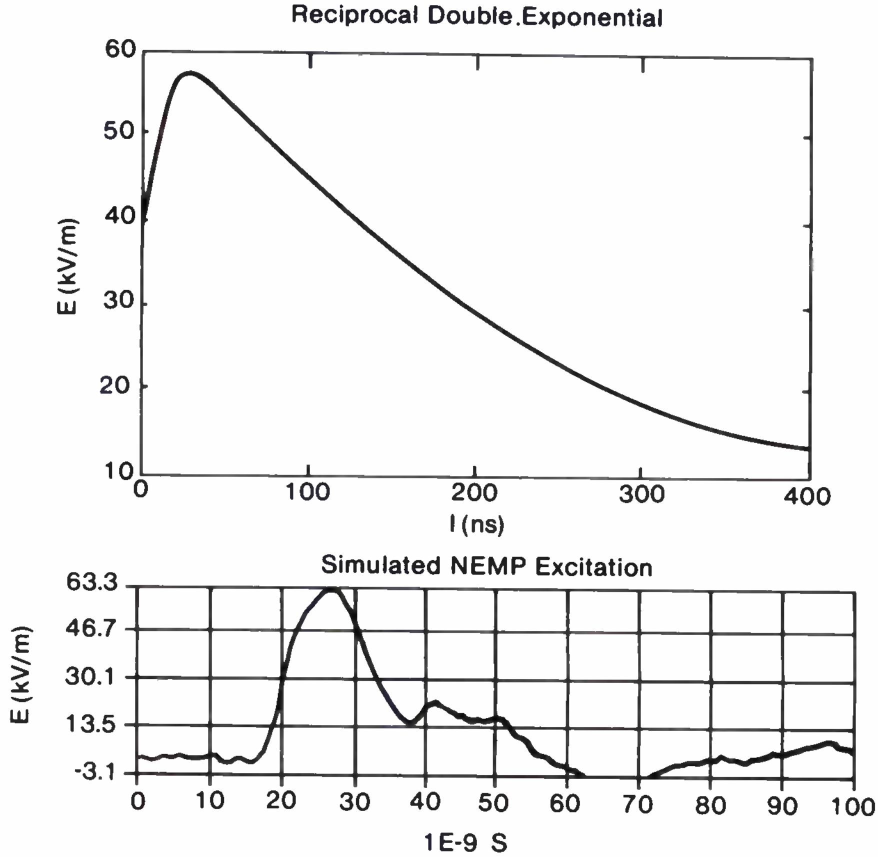

Figure 1. Reciprocal Double Exponential and Simulated NEMP Input Waveforms

The EMP from a severe lightning strike is perhaps one percent the strength of an EMP from a nuclear device. Since above ground testing has been prohibited for several years, the next best thing for studying the effects of nuclear EMP is to study those of lightning and then scale up the results. Triggered lightning experiments have been conducted in New Mexico and Florida for several years in order to place instrumentation directly at the site of a lightning strike. To trigger a lightning strike, the electrostatic field intensity is monitored and when it reaches sufficient strength, a rocket trailing a thin wire is fired upward. At just a few hundred meters altitude, the wire starts a conduction path. The wire is immediately vaporized and a lightning strike is triggered. Instrumentation and photography which is highly sophisticated then record the event for analysis.

The outcome of all this research has been to provide a wealth of information previously unavailable. This has led to studies as to the sources of atmospheric electrical energy. While much remains to be learned, more plausible theories have been put forward and, of course, these theories are the stepping stones to acquiring the facts explaining all the various processes. It must be understood that much of what will be put forth here is still theoretical and further research may modify or even throw some things out entirely. Lightning and static electricity are the effects or symptoms of atmospheric electricity, but lets look at the causes and sources of it. The existence of ions in the atmosphere is the fundamental reason for atmospheric electricity. An absence of ions would mean zero electric field in the atmosphere and most probably no thunderstorms or lightning.(1) Where do all these ions and energy come from? The sources are, in probable order of importance: (See Fig. 2)

Sources of Atmospheric Electricity

1. Cosmic Rays
2. Solar Wind
3. Natural Radioactive Decay
4. Static Electricity
5. Electromagnetic Generation

Figure 2. Sources of Atmospheric Electricity

1) COSMIC RAYS. These are charged particles, commonly known as ions, travelling at speeds approaching that of light. They are emitted by all of radiating bodies in space. Most of them expend their energy in penetrating the envelope of air surrounding the earth, creating more ions by colliding with air atoms and molecules. However, many penetrate to the surface of the earth, travelling through buildings and our bodies before they get there. One high energy particle may create up to one billion pairs of ions in the process, many of which will become atmospheric electricity.

2) SOLAR WIND. Charged particles from the sun are also continuously bombarding the earth's atmosphere. Since about half of the earth's surface is always exposed to the sun, this causes variations from day to night. Because they travel at only about 200 to 500 miles per second compared to the cosmic particles travelling at near the speed of light, their effect on air atoms and molecules is much less, but still significant. Of interest is the fact that because of their relatively slow speed, it takes from two to six days for them to travel from the sun to earth, rather than the eight minutes that light takes for the same trip. They are lumbering along at only a fraction of one percent of the speed of light. It is solar energy and not the solar wind that causes the atmospheric electricity to accumulate into giant charged cells called thunderheads from which lightning comes.

3) NATURAL RADIOACTIVE DECAY. This is the natural disintegration of radioactive elements, and in the process, some air molecules are ionized near the surface of the earth. One of the results is Radon gas which has become of concern recently as a health hazard. Radon gas can be found almost everywhere in varying concentrations. The concern is that some homes contain it in dangerously high concentrations.

4) STATIC ELECTRICITY. This is usually generated by the interaction of moving air and the earth. A hot, dry wind can generate static electricity. The spark between your hand and a doorknob is probably of little significance in the overall picture.

5) ELECTROMAGNETIC GENERATION. Because the earth is enveloped by a magnetic field, air molecules moving through the field may generate small quantities of electricity. Since the atmosphere moves along with the surface of the earth, there is little relative motion between the air molecules and the magnetic field and therefore little generation.

You may be surprised to learn that the last two effects contribute a minimal amount to atmospheric electricity, but such is the case. The combined effects of cosmic rays and solar wind account for the bulk of atmospheric electrical energy.

As mentioned at the beginning of this paper, atmospheric electricity is always present even when no thunderstorms are around. It is manifested during what can be described as "fine weather conditions", in the form of a voltage differentail between the surface of the earth and the ionosphere of 300,000 to 400,000 volts. This gradient is non-linear and near the earth may be about 150 volts per meter of elevation. Normally, the earth is negative with respect to the ionosphere and ions flow between the the two. Because there are fewer free ions near the earth than the ionosphere, the volts per meter value is probably greater because of the lower effective conductivity of the air. To express it another way, if the air is considered a semi-insulator or perhaps a semi-conductor with the free ions being the current carriers, air with more ions in it has lower resistance, and hence the voltage across a specific distance would be less. Fig. 3 shows a chain of resistors of changing values. Note that the voltage drop per meter is greater near the surface of the earth because of the lower conductivity (which is, of course, the reciprocal of resistance). The equivalent in atmospheric conductivity is to space out the iso-electric lines which represent voltages of equal steps. This may seem strange to broadcast engineers, as air is normally considered a good insulator and at relatively low voltages, ie, the normal operating voltages within our equipment, it is. But most of us have seen arcing in the process of tuning a capacitor, which indicates that air, while being a good insulator, is not perfect. We also know that at high elevation sites, transmitters and antennas are derated in power capacities because various problems known as corona and pluming will occur from FM and TV antennas and sometimes within transmitters. We know two things about high altitude sites; the air pressure is lower and the ion concentration is higher. It is perhaps the combination of both these effects that contribute to the need to derate equipment, but the increased ion content of the air at high elevations is definitely a factor. A greater ion concentration means increased conductivity and a lower ionization potential of the air.

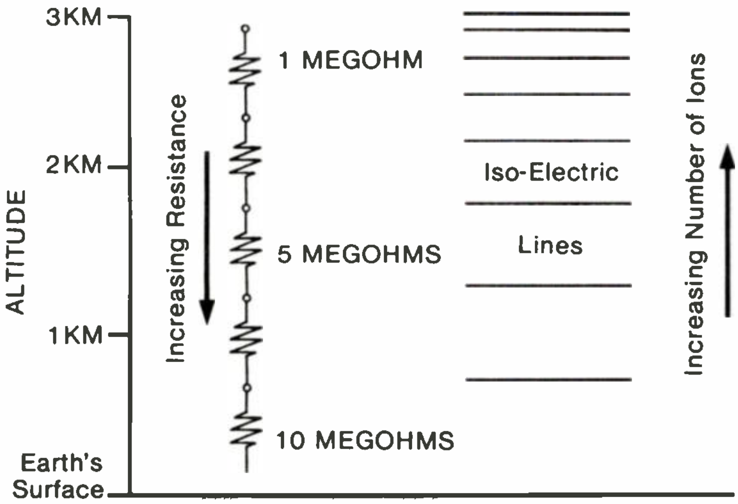

Figure 3. Atmospheric Conductivity

As we all know, we don't always have fine weather conditions. Sometimes storm clouds approach and there may be lightning and possible damage to a tower or other structure. Thermodynamic activity in the cloud causes it to become a powerfully charged cell, usually negative on bottom and positive on top. (Fig. 4) This causes a huge distortion in the voltage gradient and in fact, the polarity inverts, the earth becoming positive in reference to the cloud bottom. The voltage gradient increases to large values, sometimes exceeding ten kilovolts per meter of elevation. If you've ever experienced a tingling sensation under stormy conditions, its because of the high DC voltage gradient. Static electricity may be noted on anything not grounded. In effect, a large capacitor has formed with its plates being the surface of the earth and the bottom of the cloud. Anything within this field is subjected to its effects. The charge on the capacitor may be from 10 to 100 million volts or more, however, when the upper value is approached, the conditions are ripe for lightning discharge. If the charge can be dissipated by some form of conduction so that the voltage is reduced, the chance of lightning is lessened.

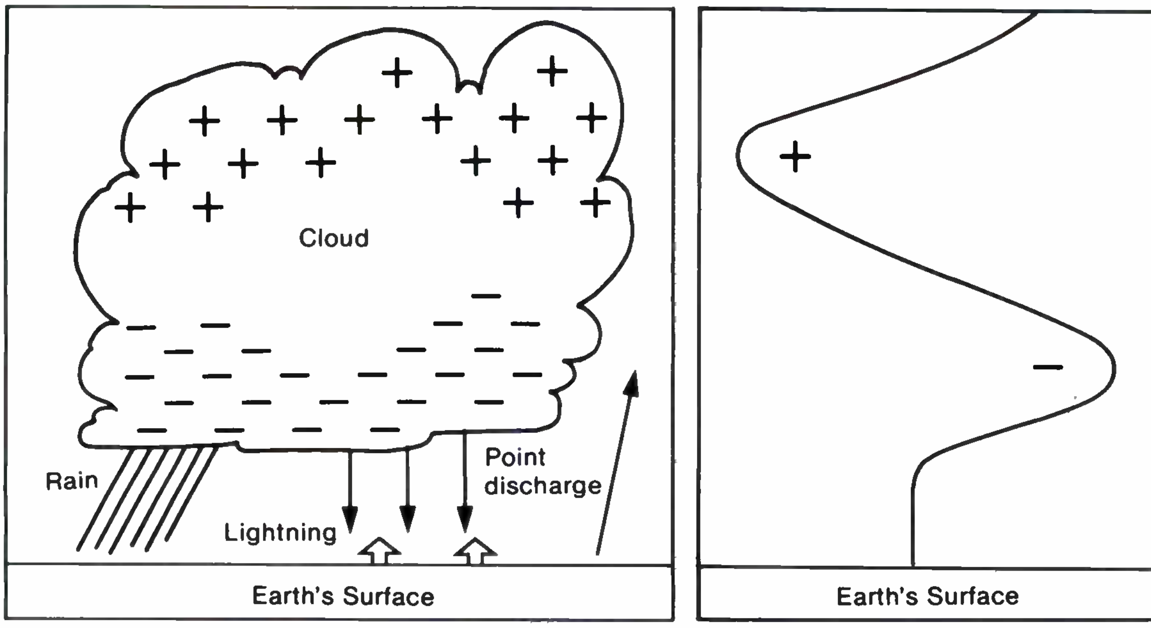

Figure 4. Storm Cell

Lets look now at a natural phenomenon called point discharge. This occurs continuously from grass blades, trees and any tall structure attempting to dissipate the electrostatic field, whether during fair weather or stormy. (Fig. 5) As far back as 1927, measurements were made by insulating a tree from ground and measuring the point discharge current. The inversion in polarity from fair weather to stormy weather was also noted. Trees are struck by lightning because they sometimes do not have adequate point discharge dissipation capacity and a flare or upward leader may occur which leads to the strike. There is a tree called the Osage Orange or Bois d'Arc which is planted in hedgerows in Kansas and Nebraska and which the old timers claim never have lightning strikes on or near them. The old timers probably do not know about point discharge, but the fact is these trees are filled with very long, sharp thorns so there may be a basis for their observations.

In the nineteenth century and the early part of the the twentieth, many experiments were conducted into the ionization potential of the air. A formula was developed which stated that around a conductive sphere, the ionizing potential was about 75 kilovolts per centimeter of radius. Imagine a ball bearing of two cm in diameter, which is a little less than an inch. A potential of at least 75,000 volts would be required to cause the air around it to ionize. Then imagine the dimensions of the members of your tower and antennas. Large voltages are necessary to cause conduction by ionization around them, and paint will increase the voltage required, also.

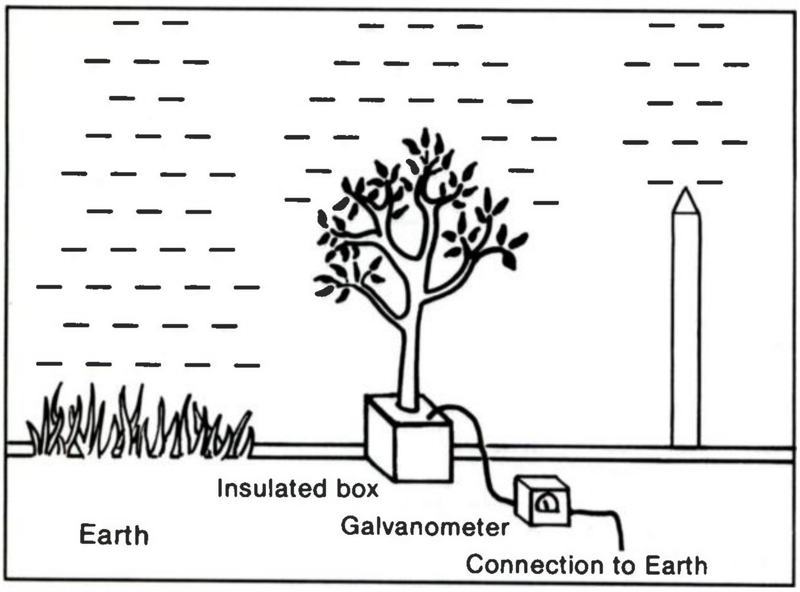

Figure 5. Point Discharge from Grass, Trees and Tower

Large copper balls were used to cause discharges called artificial lightning at higher and higher voltages. It did not take too large a ball to create discharges up in the millions of volts and that is what scientists and engineers such as Tesla did. Conversely, a very small radius would cause ionization at small voltages. Franklin, more than two centuries ago, noted what he called a "silent current" from a sharp pointed rod. Lightning rods sold door-to-door many years ago often combined a sharp point above a copper ball. The copper ball may have provided ornamentation, but the sharp point was the only part that would have reduced lightning damage. Worth noting also is the relative areas of a sharp point and a round ball. When the voltage finally reaches the value to start ionization of the surrounding air, the surface area of the ball has considerable storage capacity which can cause a flare, which may lead to a lightning strike. Conversely, the surface area of a sharp point is very small with little capacity for a sudden discharge when the ionizing potential is reached.

A table showing the balance of electrical currents illustrates the share that point discharge contributes. (Fig. 6) These apply to the entire earth, and values are in coulombs per square kilometer. Note that point discharge performs up to five-sixths of the transfer to earth. (2) Lightning and its damage might be about six times worse were it not for natural point discharge. It is estimated that only about ten percent of lightning is "cloud to ground" and about 90 percent is "intra-cloud". Were it not for this, again lightning damage would be much worse than it is. The reason for the large amount of intra-cloud discharge is the enormous voltage difference that builds up between the top and bottom of the storm cloud, and also the fact that the region between may be more highly ionized than the region between the cloud bottom and the earth.

	From Earth	To Earth
Fair weather ionic current	90	0
Precipitation	30	0
Point discharge	0	100
Lightning	0	20

Figure 6. Estimated Balance Sheet Showing Yearly Electricity Transfer for the Entire Globe, Measured in Coulombs Per Square Kilometer

While most lightning strikes are negative, ie, the cloud bottom with respect to the earth is negative, positive strikes sometimes occur and have been estimated to carry about ten times the current (200 kiloamps or more) of a negative strike. They are sometimes referred to as "hot strikes" and can cause phenomenal damage. They can occur in the winter and may happen as an after effect of a particularly active storm, when the negative lower portion of the cloud becomes depleted from many strikes. This lower negative portion may have functioned as a screen or shield between the earth and the upper, positively charged part of the cloud. When it has become depleted, the shield is removed, exposing the earth to the massive charge in the upper cloud containing charges of perhaps 500 million volts or more. If a strike occurs between this charge and earth, it has much more capacity for damage than a negative one.

SUMMARY

Previous conceptions of air being of only gaseous composition with vaporized water thrown in and thunderstorms being the generators of atmospheric electricity are now known to be incorrect. Stellar events that occurred light years ago spray our earth and its atmosphere with atoms that have been stripped of most or all of their electrons and in the process of crashing in, collide with air molecules which are knocked apart creating billions more ion pairs each second. The result may be compared to the acid electrolyte within every automobile battery, a solution of a fluid containing highly mobile ions which form the current that cranks the engine. The air is a fluid likewise, containing highly mobile ions, these ions being both the particles from space and the billions of ion pairs that are formed from the collisions of these particles and the air molecules of our atmosphere. Even though their average life spans are only about 100 seconds near the earth, they are constantly being replenished from deep space. Many re-combine to form neutral atoms and molecules, but many more are in motion creating electric currents between the earth and the ionosphere. Others are caught up in the thermodynamic cycle of storm cloud formation and contribute to the sudden discharge known as lightning.

Visualizing this process is difficult as engineers and technicians are trained in circuitry that consists of tangible things, such as copper conductors and electronic components. Looking out over a grassy field it is difficult to imagine that each cubic meter of air contains about one billion pairs of ions, that although they only have a life of about one hundred seconds, they are being produced at the rate of ten million pairs per second because of cosmic energy. It may also be difficult to imagine that point discharge is causing tiny currents to flow from the grass blades up toward the ionosphere, and finally, if one could see these ions it would seem like a dense fog as they are spaced only about one millimeter apart. But such is the nature of atmospheric electricity. Why can't we gather and use all this electricity? Even though the currents earthwide total some 2,000 amperes, the fairweather current for one square kilometer amounts to less than ten microamperes. Elevating a long wire with a baloon to take advantage of the electrostatic field has allowed experimenters to run tiny motors from atmospheric electricity, but when compared to solar energy collection, there is only about one billionth the energy per unit area at the surface of the earth.

As we gain more knowledge and understanding of lightning from all the research that is going forward, we will be able to better adapt present and future technology to the solution of long-standing atmospheric electricity problems.

ACKNOWLEDGEMENTS: Invaluable assistance was received from Richard Ives, PhD, Physics and Mr. Lars Wahlin, author of one of the references listed.

REFERENCES:

1. L. Wahlin, "Atmospheric Electrostatics." Research Studies Press Ltd. 1986

2. C. K. Adams, "Nature's Electricity." Tab Books, Inc. 1987

SURGE PROTECTION AND GROUNDING METHODS FOR AM BROADCAST TRANSMITTER SITES

By John Schneider, Pres.
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AM broadcast transmitters are potentially more vulnerable to damage by surges than most other kinds of industrial electronics equipment. This is because, in addition to surges induced through AC mains connections, they are susceptible to damage from lightning strikes coupled from the antenna. With the recent proliferation of solid state transmitter designs, and the virtual elimination of ancillary tube equipment at a transmitter facility, it is becoming increasingly important for the broadcast engineer to understand methods of surge protection and how to integrate these with proper grounding techniques.

Lightning Surges:

A lightning strike will take the path of least resistance on its way to ground, and a broadcast tower becomes an excellent pathway, by nature of the fact that it is an excellent conductor and is typically the tallest object in the vicinity. Some stations have been successful in avoiding lightning strikes through the use of dissipation arrays on their towers. While an effective technique, this can be costly, and is out of the price range of many stations. Rather than trying to modify natural phenomena, a practical approach can often be to provide an attractive and direct path to ground for lightning currents, and take steps to make paths through sensitive equipment as unattractive as possible. To do this requires some understanding of how lightning-induced currents will behave in a conductor.

A typical strike can have a maximum current of 20,000 Amperes. This current will have a rise time of 5 u/secs., and will take another 40 u/secs. to decay to half amplitude. Because the earth has a measurable resistance, it is impossible to entice all of this current directly into the ground. Typical installations have an earth resistance of from 5 ohms to a few hundred ohms. Also, there are many paths to ground at the typical site, each having its own impedance. Thus, the path to ground will resemble a parallel resistor network, with different currents flowing in each leg.

Presume a ground resistance of 50 ohms. Ohms law tells us that with a current of 20,000 amps there would be one million volts peak voltage drop between the tower ground system and true earth. This voltage will be seen at the transmitter on the shield of the coax, and some of it will try to find a path to earth ground through the transmitter. Further, any current flowing in the coax shield will induce current into the center conductor, which will appear at the transmitter's output network. This is a frequent cause of lightning-induced arcing inside a transmitter.

Developing An Earth Grounding System:

The most effective means of protection available to the broadcaster is through the use of a well planned grounding and bonding system. This requires treating the entire facility as a system, and thinking in terms of planned surge paths to a good earth ground. Proper grounding has a side benefit: reduced RFI and noise problems in equipment, which are often caused by differences in potential between two supposedly equal "grounds".

The primary objective of a grounding system should be to make a direct connection to earth ground, as low in impedance as possible, and then to make all other routes to ground through sensitive equipment as high as possible.

Be aware that there are two separate types of grounding at a transmitter plant: RF grounding and earth grounding. While some conductors may serve both functions, the facility should be analyzed separately with regard to its effectiveness for earth grounds. For instance, it should not be assumed that a tower is grounded adequately for lightning protection because it is connected to a radial ground system. Except in the wettest of sites, the RF ground system does not make contact with the water table. If possible, ground rods should be driven into the soil to at least this depth and bonded to the radial system.

It should also not be assumed that a single ground rod will be an adequate lightning ground -- in most cases it is not. Four or more ground rods spaced at least 10 ft. apart around the base of a tower and driven at least 10 ft. in the ground would be more appropriate in normal soil conditions.

Some Bonding Principles:

Conductors to earth ground should be short to be effective. The inductance of a conductor is the major factor its in impedance to a surge, and increases dramatically with length of the conductor. For instance, consider a conductor 10 meters in length of #6 AWG copper wire. It has a DC resistance of .013 ohms, and an inductance of 10 uH. For a 1000 Amp surge with a one u/sec. rise time, the resistive voltage drop is 13 volts, but the reactive voltage drop is 10,000 volts. Further, any bend in the conductor will greatly increase the inductance, and further decreases its effectiveness. A 90-degree bend is electrically equivalent to a quarter-turn coil, and the sharper the bend, the greater will be the inductance.

A large O.D. conductor should be used for all significant current-carrying connections. Four inch or larger copper strap, or #2 AWG solid wire is recommended. Because of the fast rise times, a lightning surge will behave more like AC than DC, and so the skin effect of the conductor becomes a factor. Use multiple conductors from reference point to ground where practical. The use of multiple conductors lowers the impedance of the overall network, by decreasing the current flow and hence the voltage drop in each leg.

Use silver soldering or cadwelding for all connections exposed to the weather. Corrosion will dramatically increase the resistance of a connection.

If there are multiple earth ground connections, they should be bonded together to equalize any potential between them. These bonding routes should not flow through sensitive equipment. If local electrical code or other factors prevent direct bonding, then bond through an appropriate surge arrester to allow it to conduct only during a surge.

Dealing With Problem Soil Conditions:

Sometimes the high earth resistance of certain sites requires special attention. Sandy or rocky soil has a high inherent resistance. Mountain top sites are a particular problem. Frozen soils or permafrost have poor conductivity as well.

One new method that can enhance the earth contact in poor soil conditions is a chemically charged ground rod. This is a copper pipe filled with earth salts or other chemicals. It has breather holes above the ground level to let air into the rod.

Moisture condenses inside and travels down the length of the rod, picking up the salts along the way. It is finally released into the soil through another set of holes at the bottom of the rod. The release of these fluids at the base of the rod dramatically improves the rod's contact with the soil.

Another effective method makes use of a building's concrete foundation or a tower's concrete base pier to make the ground connection. Called a "Ufer" ground, this method was developed for the Army in World War II. The steel rebar of a foundation is securely welded together before the concrete is poured, and a conductor is brought outside for bonding to the equipment. The concrete acts as an excellent conductor between the rebar and the soil. Welding of the rebar is a must, to prevent the concrete from splitting or cracking during a lightning strike.

At the Tower:

The following are some of the important steps to be taken at the base of a tower to maximize protection against a lightning strike:

Weld each tower section together for at least one of the legs, to assure good lightning path to ground, as well as good RF continuity.

Adjust the ball gap at base of tower, located across base insulator. Make sure the contacts are close enough to be effective -- just beyond the point of flashover with modulation peaks. Typically, they should be set for 2/100 inch per 1000 volts peak. Orient the terminals horizontally to keep rain water out of the gap. Keep the terminals clean of corrosion, which would otherwise increase the resistance of the gap.

There should be an inductive loop in the connection from the A.T.U. to the tower. This creates an inductive reactance to the path, and encourage the surge to jump across the spark gap. This loop is usually made from copper tubing which has been wrapped around a cylindrical form of appropriate size for two or three turns. It helps to fill the tubing with sand first, to prevent it from collapsing as it bends. Run the tower lighting conductors, if any, through the tubing to give equal inductive reactance to both paths.

Install an air spark gap to ground at the output of the tuning unit. There should also be a static drain choke in parallel across this same path to drain off wind-generated static. This is an inductor consisting of many turns of #12 wire that is connected across the base of an insulated tower.

It presents a high impedance to RF and yet a direct DC path to ground for accumulated static. A 100K ohm non-inductive resistor can also be used, about 200 watts. While effective for static electricity, these chokes cannot handle heavy currents, and typically will be destroyed by a direct lightning hit. They are readily available and inexpensive, and so it is a good idea to keep a spare on the shelf.

The ground radial system should be augmented by four or more ground rods driven ten feet apart. Also install a ground stake at each guy wire anchor, and bond it to a ground radial.

If there are any antennas mounted on the tower (FM, STL, etc.) their coaxial shield needs to be bonded to the tower at several locations, plus grounded to the tower base below isocoupler. If the antenna is for a receiver or low power transmitter, it is also appropriate to install any of several makes of coaxial surge arresters, which connect in series with the line and protect the center conductor.

All of these methods will provide a low impedance path to ground at the tower, and inhibit surges going to the transmitter. Despite these steps however, some currents will still flow to the transmitter through the outer shield of the coax, the tower lighting AC supply, and other paths. Further protection will be needed at the transmitter building.

Single Point Grounding:

In the transmitter building, it is important to establish one master ground point, and tie all inside ground connections to it with a branching system. Then, solidly ground this point to a ground rod network outside the building, using short, direct connections. It is most important that each piece of equipment offers no more than one path to ground.

An excellent location for this reference grounding point is at the AC service entrance to the building, where it can be bonded to the power company's ground (as allowed by local power codes). Bring all coaxial cables to this point directly from their entrance into the building, and securely ground the shields before routing them to the transmitter or other equipment. Surges will tend to enter the building via either the coaxial cable or power lines. The use of this method will provide a direct path to ground for those surges, one which will not flow through any of the equipment inside the building.

Install Surge Protectors:

AC line surge arresters, if used, should be installed at this point between the AC service entrance and the reference ground. The arrester should be located as close as possible to the primary AC service entrance. This serves two functions:

1. It short-circuits any surge originating on power lines before it gets to the transmitter.

2. The AC neutral presents a low impedance path to ground for a lightning strike on the tower, due to its large conductor size and distribution to many ground connections upstream from the service entrance.

The surge arrester should be sized sufficiently to handle multiple 20 Amp surges without damage.

Most surge arresters install in shunt across the protected device to ground. In other words, they will share the surge with the protected device. In order to be effective, the path impedance of the arrester must be made lower than the path through the load. It is of primary importance to keep the lead impedance low between the arrester and ground, as the lead impedance and device impedance will add. The better surge arresters will also provide an inductance element in series with the power line itself, to increase the impedance of the path through the equipment. If this kind of arrester is not being used, an inductance can be added externally.

Adding A Series Impedance:

Add a series impedance on the coaxial cable or power lines between the ground connection and the equipment. This is accomplished by passing the conductors through one or more ferrite cores or torroids. This will increase the impedance of this path to ground relative to the preferred path, and make the path through the equipment look unattractive to a surge. It can also slow the destructive rise time of the surge. If using this method, it is essential to pass all conductors through the same torroid. This will cause opposing AC currents to cancel each other and provide no impedance to the desired signal, and yet present a significant impedance to a surge. If all conductors do not pass through the core in the same direction, the ferrite will saturate and become hot. Besides dissipating power needlessly, its effectiveness against surges would then be lost. For coaxial cable, pass the entire cable through the center of a torroidal core and fasten it to the core with tie wraps or electrical tape. It is possible to find torroids of sufficient size to accommodate up to 1-5/8 inch line. For AC power lines, pass all conductors through the center of the core and, if possible, wrap one or more turns around the core, being careful to wrap all conductors in the same direction.

(The method of installing torroids on coaxial cable has not been found to be effective when the transmitter is situated right at the base of the tower. In this instance, the outer conductor is also carrying significant ground return currents to the base of the tower, and these tend to saturate the core.)

Protecting Other Equipment In The Building:

In addition to power lines and coaxial cables, the same methods described can be applied to incoming telco audio or control lines, or any other connections to the outside world. On phone lines, connect an MOV of the appropriate voltage rating between each conductor and ground, as well as leg to leg for balanced lines. These are easily added to the punch block type of service panel. Connect the panel ground to the station reference ground. After they leave the service panel, wrap the conductors through a torroid as many times as possible before they are connected to the transmitter.

The single point grounding technique can also be effective to protect multiple pieces of equipment installed in a single equipment rack. Treat the rack the same as a building, and install a panel to serve as the entrance point for all conductors entering and leaving the rack. Install surge protectors at this point in shunt to ground, and install a series impedance between the panel and the equipment. Bond this panel to the building reference ground.

Grounding The Reference Point:

Once all connections have been made to the master ground point inside the building, it must be bonded to an effective earth ground system outside the building. The same techniques described for towers can be used here. One additional method is to run a strap around the perimeter of the building, bonded to several ground rods spaced evenly around the building. In a large building with several independent systems inside, each system can have its own reference ground point established, connected at the closest convenient point to the perimeter ground. This is preferable because it avoids long runs of ground.strap inside the building. It is important that ground connections between these systems be eliminated, and torroids or surge arresters be installed on the interconnections that cannot be avoided.

Conclusion:

These methods of protection are not difficult to implement at a new facility; however, adding them to an existing installation can be more difficult. Often, facilities have grown so complex and interconnected over time that reworking them becomes a monumental task. This brings rise to the question: How much protection is enough?

This question is answered by applying the reasonable risk concept: Given the statistical occurrence of lightning in the area, and the relative susceptibility of the equipment to damage from surges, one should expend enough time and money as is required to reduce the risk of damage to tolerable levels.

Some of these methods are inexpensive and easily implemented, and they are the most important ones to consider adding to your operation. Further, if you find yourself planning a new facility, you have a golden opportunity to "do it right" the first time.

REFERENCES:

1. Booklet: "Lightning Protection for Radio Transmitter Stations"; by Nautical Electronic Laboratories, Ltd. (Nautel),Hackett's Cove, Nova Scotia. Second printing, copyright 1985.

2. "Lightning Protection for Broadcast Facilities", by Marvin Frydenlund and Frank Breeze: N.A.B. Engineering Handbook, Seventh Edition.

3. Electronic Protection Guide for Land-Based Radio Facilities by David Bodle; Joslyn Manufacturing & Supply Co., Inc., Goleta, CA.

4. "Grounding, Bonding for Transient Protection" by Henry L. Claussen: Mobile Radio Technology, August 1983.

5. Series of articles by Roger Block, Polyphaser Corp., in Mobile Radio Technology, September 1984 to September 1986.

ACKNOWLEDGEMENT:

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IMPROVED GROUNDING METHODS FOR BROADCASTERS

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Introduction

Interfacing a communications system with earth is an essential element of a broadcast system. System performance, particulary at the lower frequencies, depend on a low impedance, low resistance interface with mother earth. Further, the performance of the support systems, such as the power source, and lightning protection systems depend on a very low ground resistance to provide a safe, reliable support system. Unfortunately, many sites are located in areas where low resistances are not practical through use of conventional grounding technology.

As an example, the lightning protection system that depends on a diversionary principal such as a lightning system, very low resistances are required to protect the broadcasters investment. Resistance of 10 ohms were thought to be satisfactory; however, contemporary systems have been found to be intolerant of the resulting voltage transients. As an example, a 10 ohm ground will permit a 200Kv transient to be developed across that ground with just an average 20,000 ampere lightning strike. The 99 percentile 100,000 ampere strike would destroy everything in its path.

If the station is protected with an LEC Dissipation Array System , grounding resistance is far less important because it prevents the strike to the site. With a lightning diverter system and a high resistance ground, the broadcaster can be assured of equipment damage and/or program interruptions as a result of lightning activity in the transmitter area.

Where grounding/counterpoid systems are required for the normal operation of the system, low resistances will mean better performance. For example, the low frequency statións (such as AM stations) a better counterpoid ground means a better ground wave and a greater market. For them the ideal situation is a salt marsh; or, a bed of some highly conductive soil.

Low resistance ground interfaces for the lower system assure better performance of the Surge Protectors, the regulators and line filters; and therefore your system.

In summary, low resistance grounds are essential to good broadcasting; and, the lower the resistance the better overall performance there is.

THE GROUNDING ELECTRODE INTERFACE MECHANISM

The grounding electrode, normally thought of as a ground rod, is a conductor, regardless of the form. It must interface with a semi-conductor, the earth. Studies have shown that this interface requires a hemispherical-shaped volume of that soil (1), the size of which is related to the size of the grounding electrode, and is more sensitive to electrode length than diameter. This can be demonstrated by measuring the change in resistance per unit distance, as a function of distance from a driven grounding rod. It is observed that the change in resistance decreases exponentially with distance from that rod. At some reasonable distance from the rod, that change in resistance will be insignificant or immeasurable.

It has been found that 90% of that change has been accomplished at a radial distance equal to about 1.1 times the length of the grounding electrode from that electrode. That value changes somewhat with the electrode diameter.

For design purposes, that change may be considered negligable. Therefore -- each earth electrode may be considered to require a hemisphere shaped volume of earth, about 2.2 times the electrode length in diameter to form a proper interface with earth, as illustrated by Figure 1. Table 1 presents the approximate interface hemisphere size for three rod sizes and various levels of interfacing effectiveness.

Table 1, Effective Hemisphere Size Factors

Standard 1 inch Rod of Length:	*Diameter of Hemisphere (Rt.)		
	90%	95%	99%
6 foot	11	22	112
10 foot	34	68	340
30 foot	86	172	862

FIGURE 1. THE EFFECTIVE EARTH INTERFACE HEMISPHERE

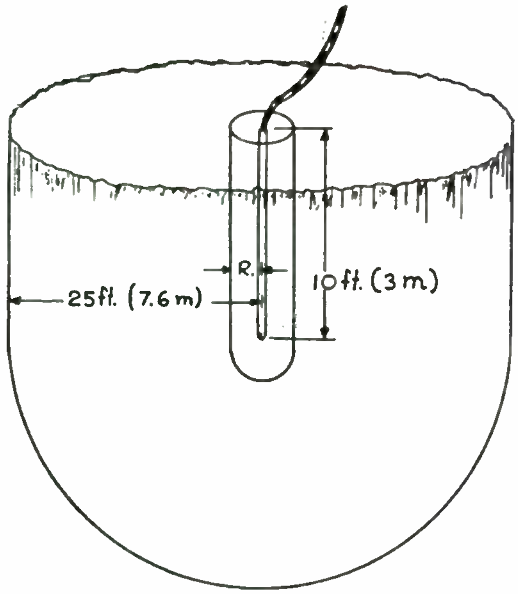


FIGURE 2. EFFECT OF OVERLAPPING EARTH INTERFACE HEMISPHERES (ETH)

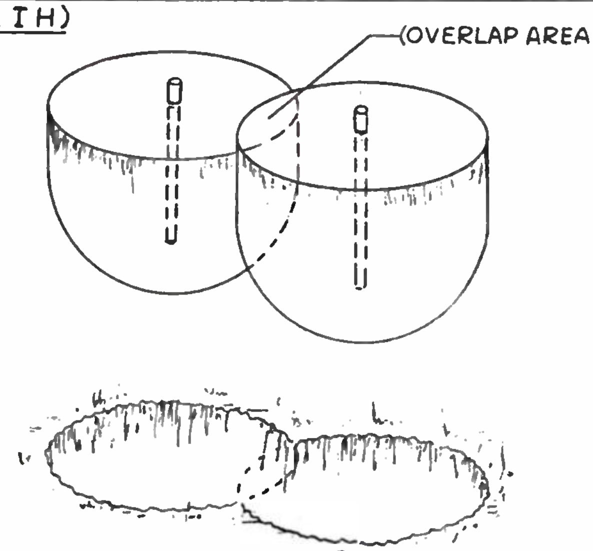


FIGURE 3. GROUNDING RESISTANCE AS A FUNCTION OF ROD LENGTH.

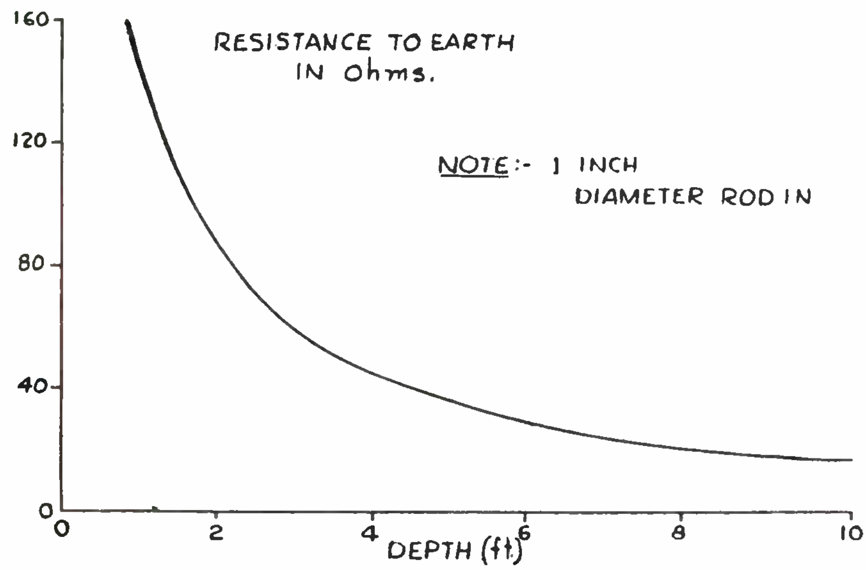
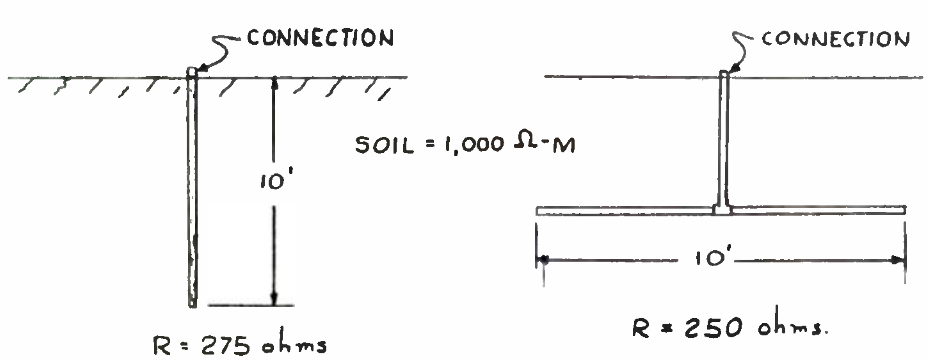


FIGURE 4. THE HORIZONTAL vs. THE VERTICAL ROD



An all too common practice is to drive these electrodes in too deep, forcing wider separation distances; or driving them too close together. As illustrated by Figure 2, when rods are too close, part of the interfacing media is lost. The resulting gain that is experienced from the addition of a second or subsequent rods, is proportional to the interface area allowed. The rods must be properly spaced to achieve a significant lowering of the grounding resistivity. As an example, two 10 ft. rods separated by only one foot provide a resistivity approximately equivalent to that of the single 10 ft. rod.

It is also an erroneous belief that extending the rod length, perhaps to reach more conductive soils, will reduce the actual grounding resistance. The measured resistance may seem to decrease, however, that conclusion is usually the result of a poor measurement technique that is misleading the user.

NOTE: For proper measurement techniques for long rods and large areas, refer to reference 7. It should suffice to state that the longer the rod, the further the separation requirement for the measurement system.

It has been found that as a rod is driven into the earth, the return in reduced resistivity, per driven foot of rod, decreases exponentially with depth. This factor is best illustrated by Figure 3, which suggests that rods driven beyond five feet in length, in good moist soil, provide very little additional benefit. It appears that in poor soil, the optimum length is about 10 feet beyond, which seems to be a waste of time and money, for most situations. A more cost-optimized solution, usually, is to use many short rods at a closer spacing; this results in better land utilization when the correct moisture level is present.

Since short rods seem to provide more return per unit length than long rods, it would appear evident that there may be a greater return from many short rods vs. a few long rods. Since a five foot rod requires a hemisphere of 11 ft. diameter and a 10 ft. rod requires a 22 ft. diameter, over twice as many can be included in a given area. Tests have shown over a 33% reduction in resistance for the shorter rods. (3)

NOTE: Refer to the sections on soil stratification which illustrate this phenomena.

Table 1 presents some sample spacing requirements for one inch diameter rods, 10 feet long, as a function of the percent effectiveness desired from each electrode. From these data, it seems obvious that the 90 percentile spacing is the most cost effective spacing parameter.

THE USE OF HORIZONTAL RODS

Since the foregoing data seems to indicate that short rods, using the more conductive upper soil layers, provide the lowest resistance per unit foot of rod, it would seem likely that horizontal rods, or buried wires, may provide a greater return for a given investment in rods than long vertical rods; possibly even those of only 10 feet. To assess this situation, a 3/4 by 10 ft. rod, driven vertically in 1,000 ohm - meter soil was compared with the same rod layed horizontally, as in Figure 4. The results are as follows:

R Vertical = 275 ohms
R Horizontal = 308 ohms at 36 inches deep;
R Horizontal = 250 ohms when connection to surface is included.

Therefore, the horizontal rod provides a resistivity 110 percent of the vertical rod of the same length; however, when the vertical connection to the horizontal rod is included, its resistivity is ony 90 percent of the vertical rod resistivity. This generality is of course independent of soil resistivity, but fully dependent on moisture content.

THE EFFECTS OF SOIL STRATIFICATION

All of the previous data was based on the premise that the soil around the grounding electrode(s) was reasonably uniform. However, in the real world that is often not true. The norm is at least two strata within 10 feet of the surface, and sometimes more. Further, it is usually true that the upper strata is the more conductive; when, and if, there is sufficient moisture.

The influence of stratification can best be illustrated from the situations depicted by Figure 5. Six situations have been illustrated, using two reasonable conductive soil conditions, 100 ohm - meter and 500 ohm - meter soils. The following facts were identified:

1. A 3/4" by 10 ft. rod provided 32 ohms grounding resistance in all 100 ohm - meter soil;

2. That same rod in all 500 ohm - meter soil provided 150 ohms grounding resistance;

FIGURE 5. THE IMPACT OF SOIL STRATIFICATION

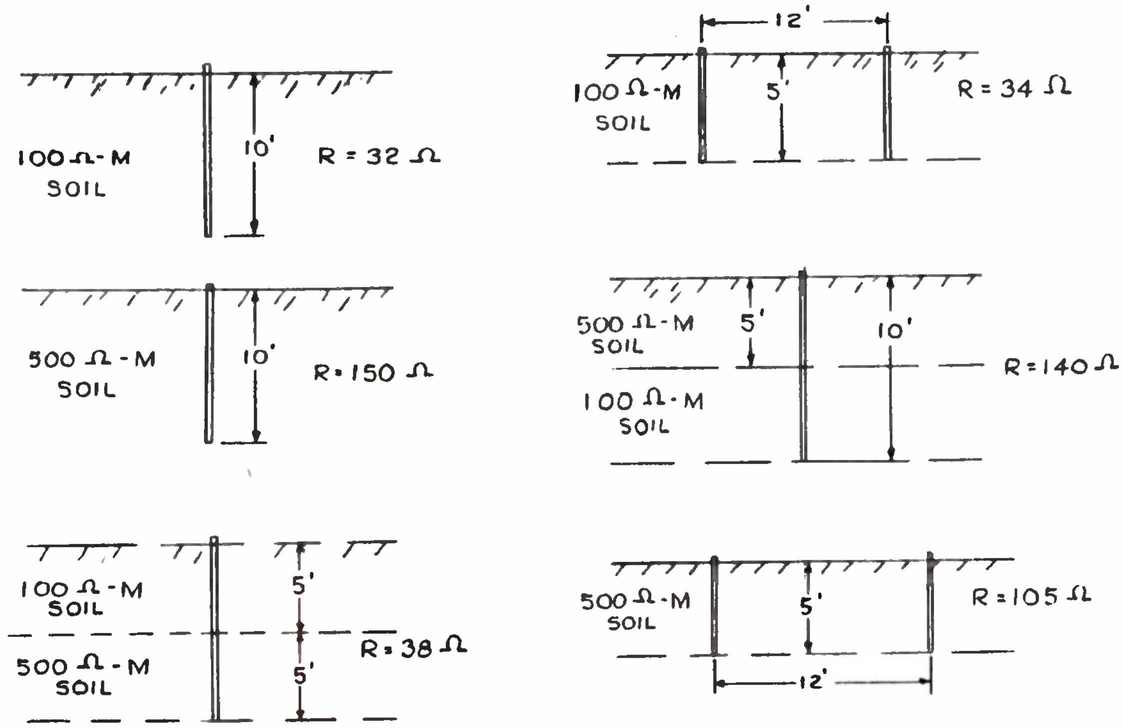


Figure 6, NUMBER OF RODS REQUIRED vs. SOIL RESISTIVITY

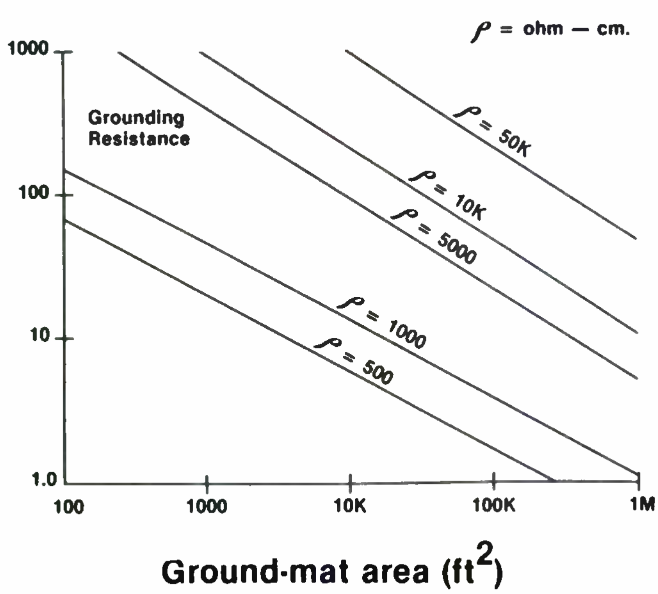
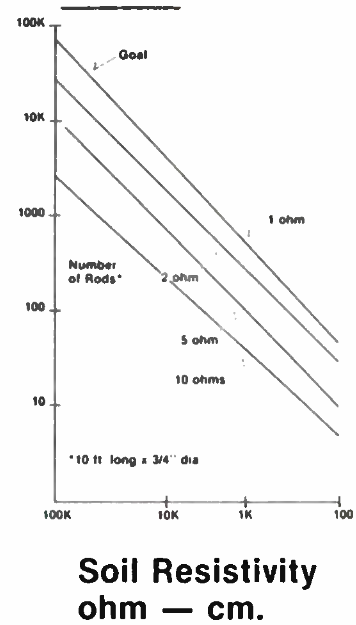


Figure 7, GROUNDING RESISTANCE FOR VARIOUS MAT AREAS



3. That same 10 ft. rod in five feet of 100 ohm - meter top soil, then five feet of 500 ohm - meter soil provided a 38 ohm ground;

4. However, that 10 ft. rod cut in half, where each piece was driven 5 ft, separated by 12 feet provided 34 ohms resistance;

5. Reversing the strata sequence with the 10 foot rod, first 5 feet of 500 ohm - meter, then 5 feet of 100 ohm - meter soil, offered 140 ohms of resistance;

6. While the two five foot rods in the 500 ohm - meter soil provided a resistance of only 105 ohms, significantly better.

The conclusion should be obvious; more often than not, the use of many short rods, which permit closer spacing, provides a lower resistance than a lesser number of longer rods, even when the lower strata is more conductive. As with any generality, there are of course exceptions. In very arid areas and dry soils, low resistance grounds are inpractical with conventional technology. These areas must be treated with moisturization.

USE OF MULTIPLE ELECTRODES

When one rod fails to provide the desired resistivity, it is common practice to resort to the use of multiple rods, in some form of rod/wire matrix. As an example, figure 6 demonstrates the result of using multiple 3/4" by 10 ft. rods, all spaced at 22 ft. in such a rod/wire ground matt. This chart indicated that about 800 rods, properly spaced, are required to achieve a one ohm ground in uniform 1,000 ohm - meter soil. The data from Figure 7, illustrates that this would require a uniform area of one million square feet; an impractical size for many applications.

DEALING WITH THE GROUNDING RESISTIVITY CONSTRAINTS

The foregoing data leads to one conclusion; i.e., there are two basic factors that limit our ability to achieve a low resistance earth interface. These are: (1) The area of land available for the grounding system; and (2) The soil resistivity within that area.

Most often, the designer is limited in the amount of real estate available for his use; he is therefore left with only one parameter to deal with: soil resistivity. Soil resistivity is known to vary significantly with composition and state. Specifically, such factors as: soil type, compactness, temperature, moisture content, and chemical or mineral content,... influence the measured value. Of these factors, two or three (depending on the site location) seem to exercise the most significant influence, the mineral content and moisture. Temperature is significant only in areas that freeze. (1)(2)(3)

Soil resistivities have been found to vary from lows of less than 5 ohm - meters to highs of over 10,000 ohm - meters (too high to measure). Areas where permafrost exists are also too high to measure accurately.

Table 2 presents a spectrum of soil resistivities by soil type. These are representative of those found in various parts of the world. The wide variations within the individual types, are due to variations in moisture content, minerals and temperatures, in that order.

From these data, it is obvious that soil resistivity can be influenced significantly through the control of these parameters. The remainder of this paper deals with controlling these parameters, and ultimately, the resistivity of a grounding system.

Table 2, A Sample Assessment of Resistivity
for Different Soil Types

Type of Soil	Resistivity in Ohm - Centimeters		
	Average	Minimum	Maximum
Filled land,ashes cinders, salt marsh	2,400	600	7,000
Top soils, loam	4,100	340	16,000
Hybrid soils	16,000	1,000	135,000
Sand & Gravel	90,000	60,000	460,000

AUTOMATED MINERAL ENHANCEMENT

It has been known for many years, that the addition of ordinary table salt (NaCl) to soil will decrease its resistivity. This is best illustrated by Figure 8, which indicates that when the moisture in the soil is at the correct level (between 4 and 12%), and 10% of that is salt, the soil resistivity can be reduced from a high of 10,000 ohm - meters to less than 100 ohm - meters.

However, as Figure 9 illustrates, time and rain exercise a degrading influence. Subsequent resalting never seems to reproduce the initial results. Eventually the soil resistivity returns to its original value. There is therefore a requirement for some form of conditioning that does not degrade with time.

To that end, during the past 12 years or so, there have been various attempts to achieve that objective. Typical of these devices, is the one illustrated by Figure 10. It is a 2 1/4" diameter copper pipe, filled with rock salt, provided with breathing holes at the top, and seepage holes at the bottom. Moisture is absorbed from the air (when available) absorbed by the salt, creating a solution that seeps out the base of the rod and conditions the soil in that area. As a result, in some areas a dramatic reduction in soil resistivities were experienced (see Figure 11). In dry areas it was significantly less effective.

Figure 8, EFFECT OF SALT ON RESISTIVITY

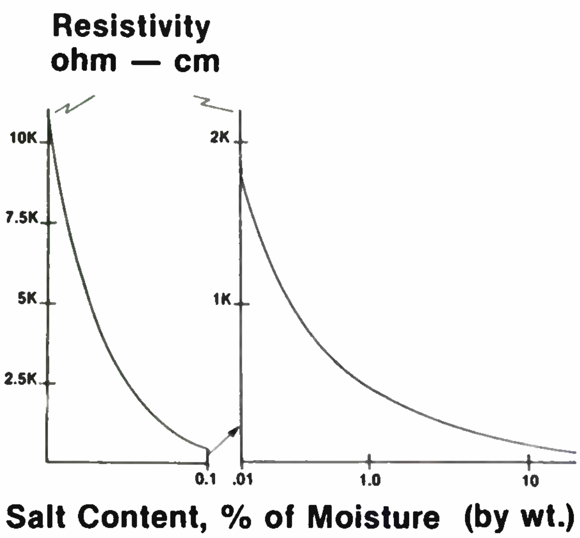


Figure 9, SOIL SALTING IMPACT ON GROUNDING RESISTANCE WITH TIME

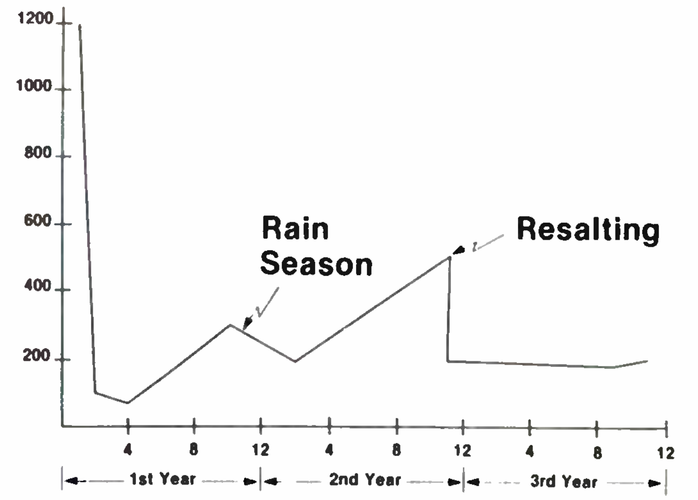


Figure 10, TYPICAL AIR BREATHING CHEMICAL ROD

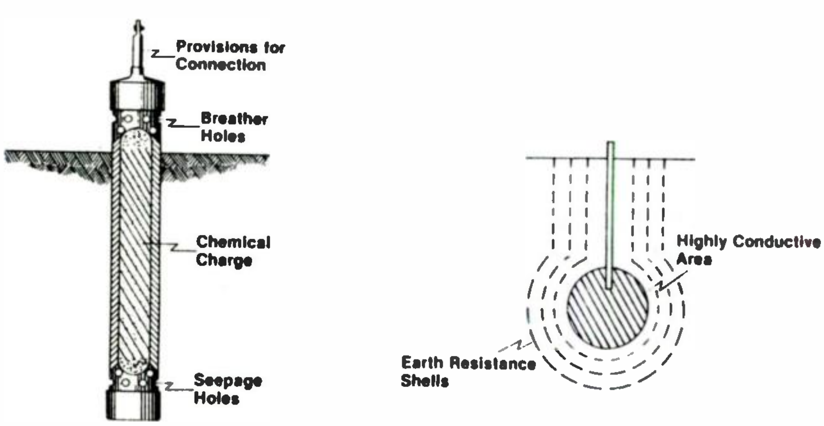
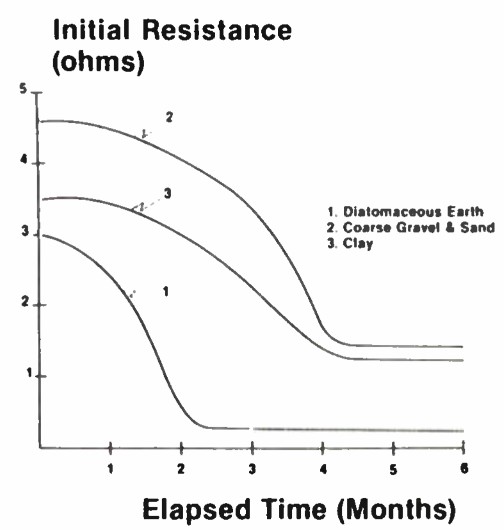


Figure 11, XIT ROD PERFORMANCE



Subsequent tests have shown that these "air breather" concepts suffer from two limitations: They require moisture laden air, and they condition the soil around the base of the rod only. From an earlier section of this paper and studies by several researchers it becomes obvious that much of the interfacing soil hemisphere must be conditioned to obtain the best possible results, and approach that which was obtained through the initial conditions illustrated back on Figure 8. In fact, it can be shown, that it is only necessary to condition the soil within the interfacing hemisphere. The soil external to it has little influence on the true resistivity of the interfacing electrode to earth.

Based on the foregoing, Lightning Eliminators and Consultants, (LEC) Inc., developed the ultimate automated soil conditioning system - The Chem-Rod. The Chem-Rod concept as illustrated by Figure 12 is provided in a variety of lengths, in both the horizontal and vertical assembly. In addition they are offered in a modularized form to provide the designer with an endless assortment of deployment options.

In contrast to other chemical rods offered, the Chem-Rod assembly conditions a much larger percentage of the soil, within the interfacing hemisphere. Figure 13 illustrates the principle of operation. Moisture is absorbed from the soil (and rain) through the ports. The delequessent metallic salts absorb the moisture and form a saturated solution of those salts.

The salt solution seeps out the many ports provided for that purpose and infiltrates the surrounding soil by Osmosis. That soil then becomes very conductive because the mineral content is increased, and its sensitivity to moisture has thereby decreased.

NOTE: The higher the mineral content in soil, the less its sensitivity to moisture content.

The resultant impact on grounding resistance for that electrode is dramatic. As illustrated by Figure 14, if the moisture content is within the required range, the resistivity to earth will be reduced by as much as 100 to 1; possibly more.

The impact of this conditioning process is best understood in light of the CR impact on the interfacing hemisphere. As previously illustrated, each grounding electrode, regardless of its character, when used to establish electrical contact with earth, requires a hemispheric volume of that earth to properly effect the interface. It is also equally true that the conductivity or resistivity of the soil <u>within</u> that hemisphere determines the resistivity of that electrode to earth.

Figure 12, CHEM-ROD CONFIGURATIONS

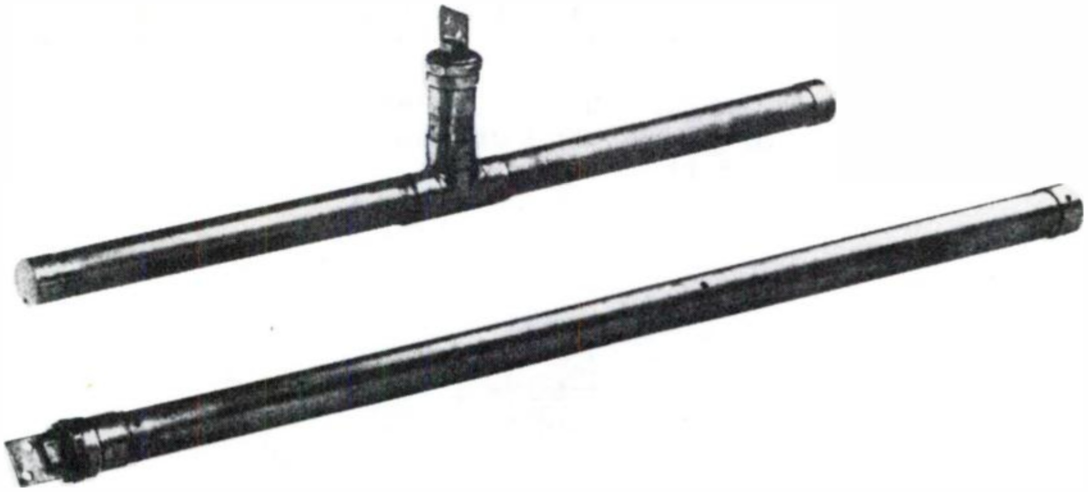


Figure 13, CHEM-ROD SOIL CONDITIONING PROCESS

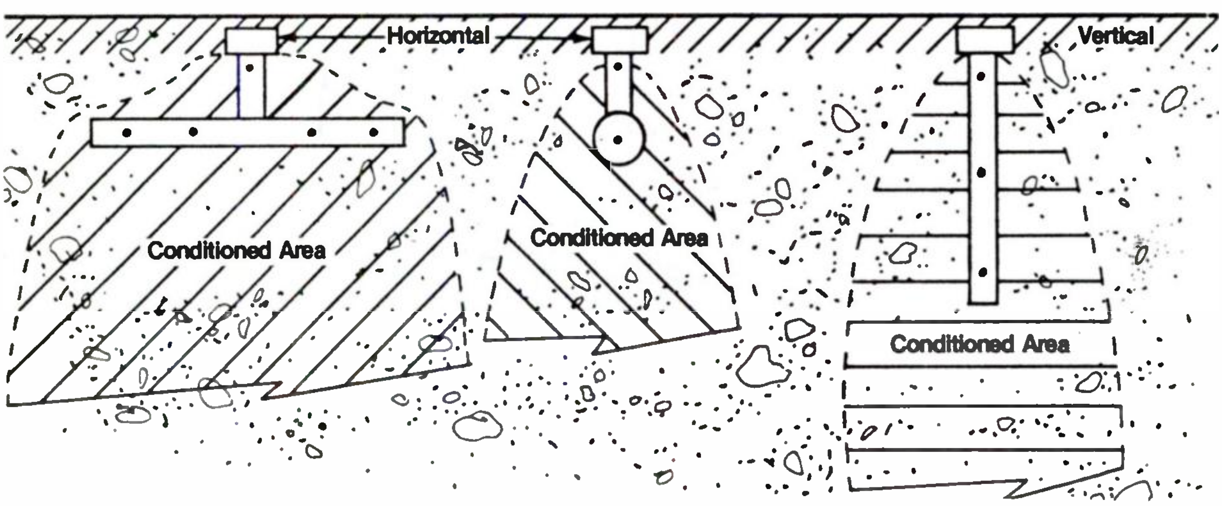
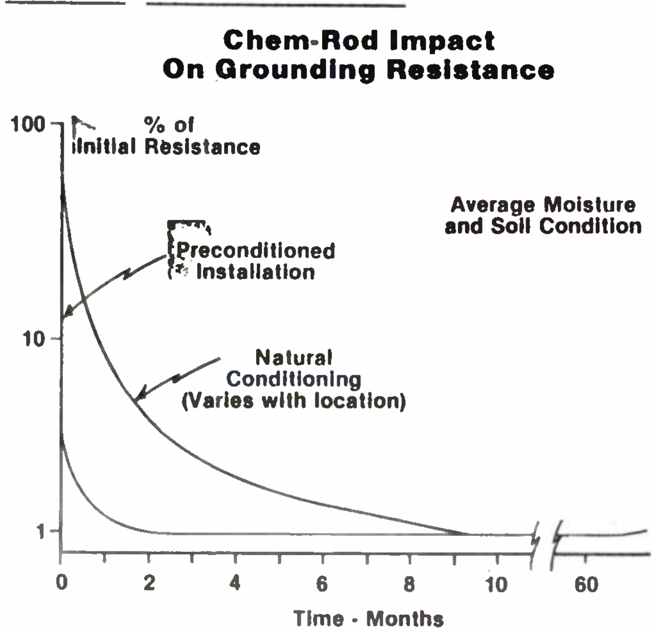
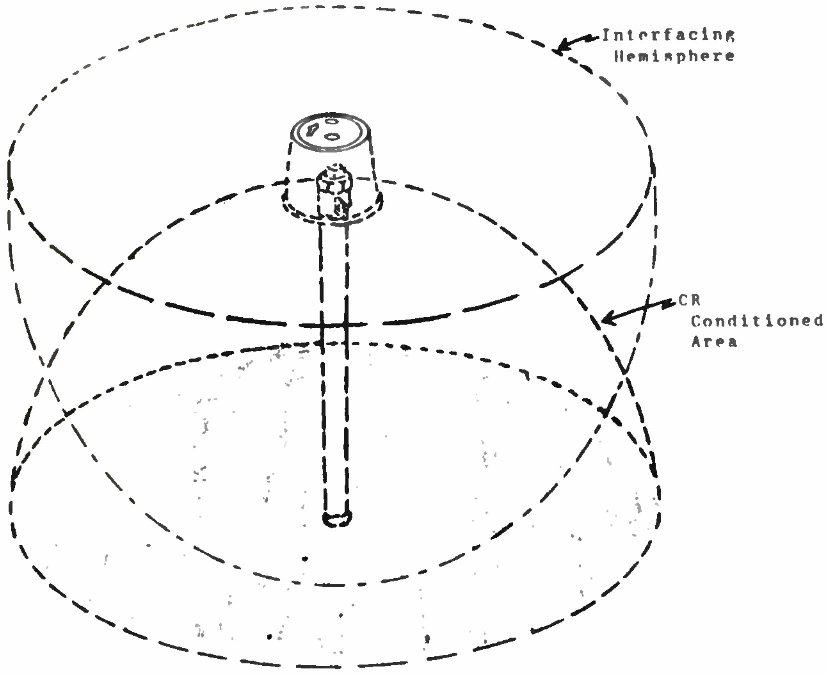


Figure 14, CHEM-ROD PERFORMANCE



Given these data, it also follows that any system that conditions the soil within that hemisphere will exercise a predominant influence on the ultimate resistivity. As illustrated by Figure 15, the Chem-Rod both conditions a major percentage of the interfacing hemisphere; and extends the effective elecetrical length of the rod.



The Fully Automated Counterpoid

Just as a desert can bloom and become a lush garden with enough water; dry high resistance soil can be made as conductive as a salt marsh - with proper treatment. It can also be transformed into an ideal counterpoid for even the AM broadcaster. All that is required is the correct amount of moisture and conductive mineral, properly distributed throughout the soil within the counterpoid interfacing area; adjusted to provide the desired conductivity.

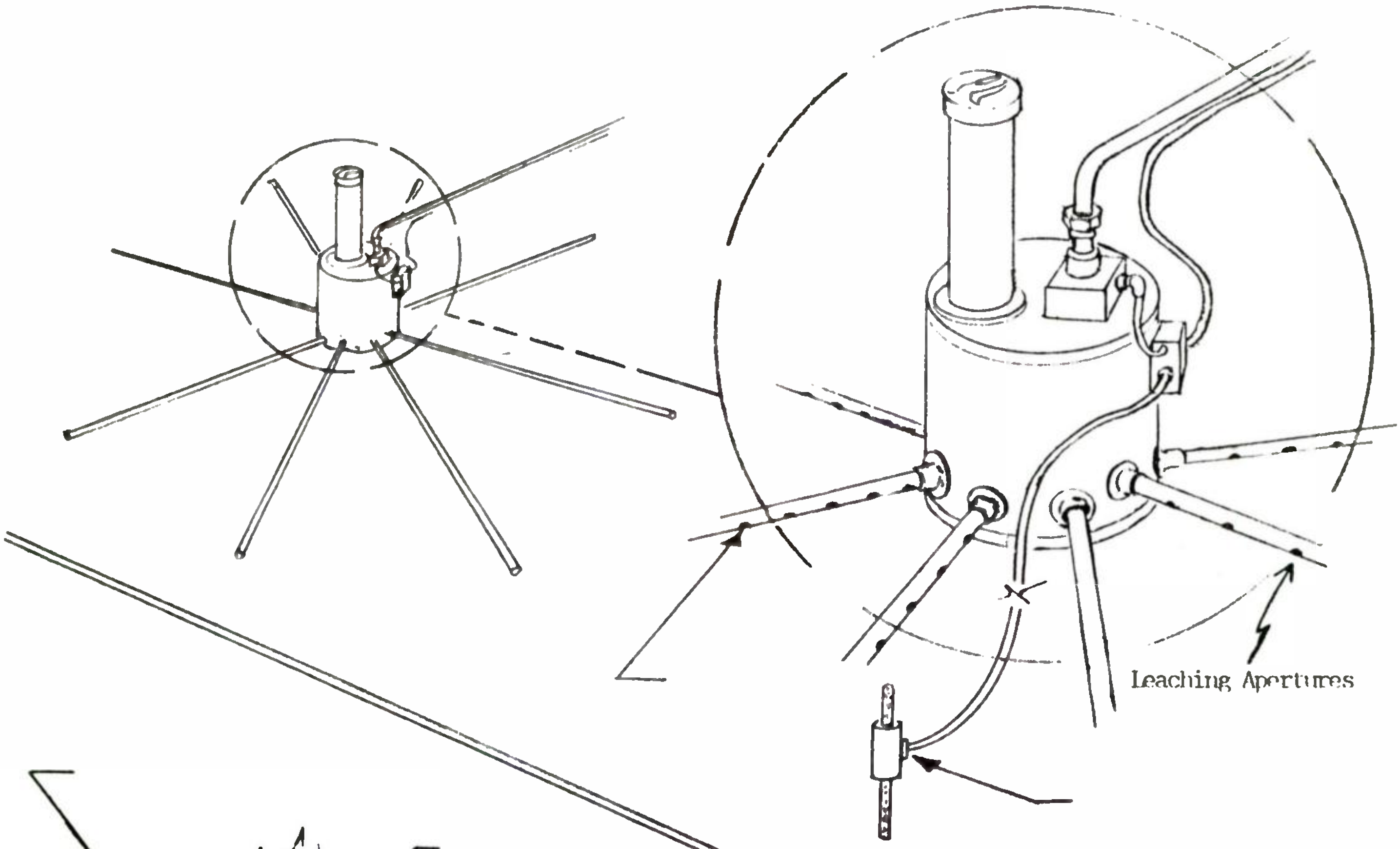
The fully automated counterpoid (or ground matt) is one that performs three functions:

1. Provides the correct amount of moisture continuously;

2. Provides and maintains the correct amount of minerals; and,

3. Provides the interfacing conductors.

Figure 16, The Fully Augmented Grounding Counterpoid



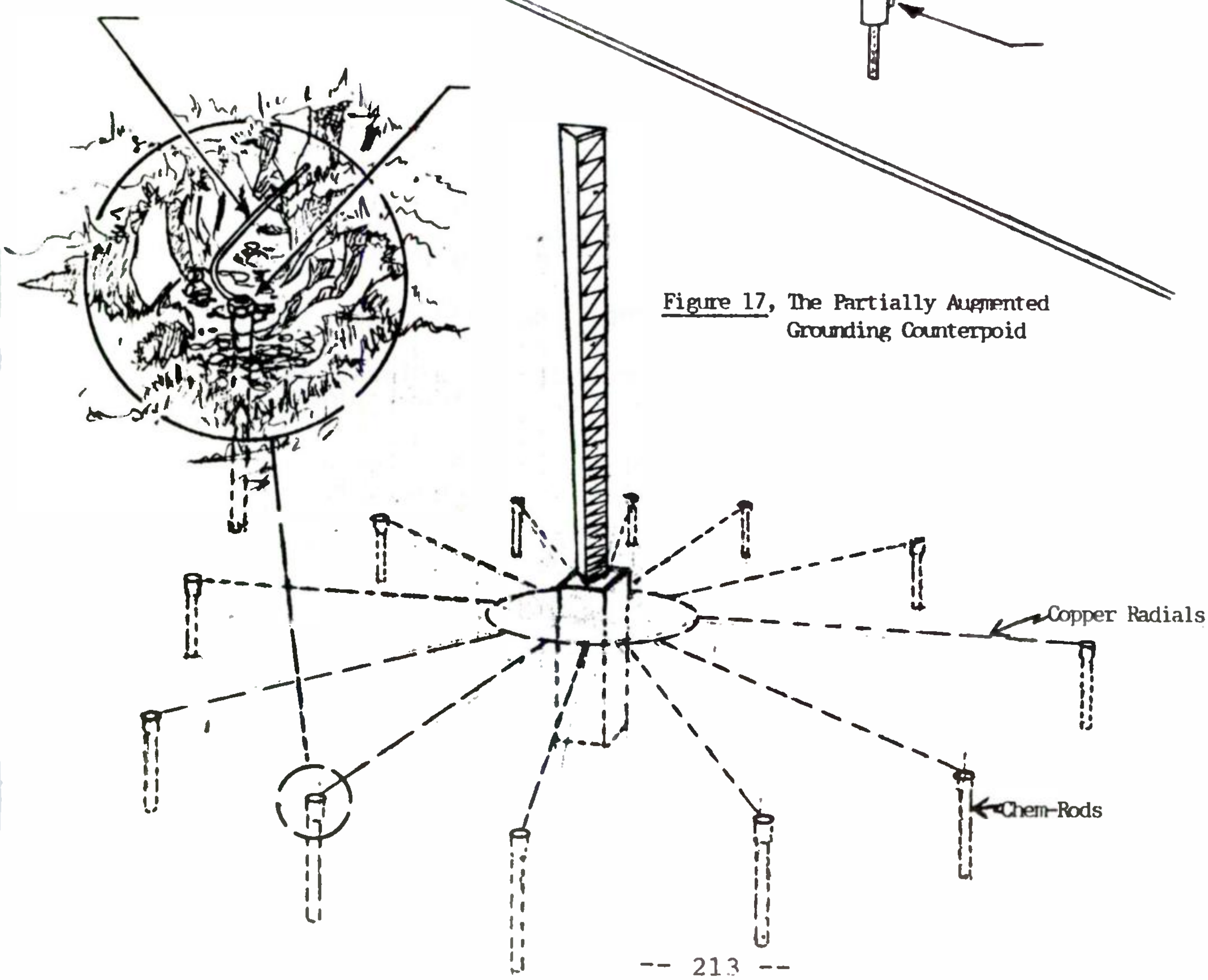


Figure 17, The Partially Augmented Grounding Counterpoid

Two such systems are illustrated by Figures 16 and 17 for which patents are pending. Figure 16 illustrates a fully automated system, where all of the functions are provided by the counterpoid system. It is a modification of the Chem-Rod system where the counterpoid conductors are made up of perforated copper tubing; much like a drip irrigation system.

The central hub provides the manifold and the source of minerals. A moisture function which injects the moisture into the manifold and thence out through the conductive tubing, with the minerals.

Figure 17 illustrates a hybrid system for use in areas where there is enough moisture, but soil of high resistance. The Chem-Rods continue to make the soil more conductive through its natural process.

In summary, low resistances can be obtained, even under "worse case" conditions, through the logical use of mineral and moisture control.

BIBLIOGRAPHY

1. Lightning Protection, J. L. Marshall, Wiley, 1973

2. IEEE Std. 142-1982, Grounding of Industrial and Commercial Power Systems, 1982

3. Tagg, G. F. Earth Resistance, London W C 2 George Newnes Limited. Tower House. Southhampton St.

4. FAA -RD- 77 - 84, Federal Aviation Administration Florida Institute of Technology Workshop on Grounding and Lightning Protection, May 1977

5. HVDC Ground Electrode Design, EPRI EL 2020, August 1981

6. Carpenter, Roy B., Grounding Concepts, Considerations and Configurations, LEC No. 85.002, December 1985

7. "Manual On Earth-Resistance Testing for the Practical Man", Biddle Instruments, Bluebell, PA April 1981

8. "Earth Resistance Tests with Three Point Vibroground", manual 18507, Associated Research Inc., Chicago, Illinois, 1972.

9. "Standard Handbook for Electrical Engineers" Funk and Carroll, Tenth Edition, McGraw Hill, 1969.

USING MANAGEMENT SCIENCE IN BROADCASTING

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Some major corporations in our industry do use management science techniques. However, most medium and small market operations do not. The normal procedure is to draw up a set of plans and go to work. The results can be equipment that arrives too early costing both storage fees and interest charges. Labor may be improperly assigned, either too much or too little for a given task, or worse yet, the correct labor may be there, but there may be no equipment or materials.

This paper looks at the use of CPM "Critical Path Method," Pert Charts and Gantt Charts in the construction of major projects in the field of broadcast engineering.

The use of CPM results in a timetable with the starting and ending dates of a task. By knowing these dates one can order equipment at the proper time and actually enter the order date as a task on the CPM chart. Proper planning results in a savings of both manpower and operating expenses. This paper explains the function of the Pert Chart, Gantt Chart, and resources calendar and how they were used to control the construction of a transmission facility project.

BACKGROUND

The first question to ask is "What is a Project?" The MBA school definition of a project is a series of related tasks performed to reach a specific objective. In English that means the tasks must be related to be considered a project. An engineer who runs around his facility fixing one emergency problem after another is not working on a project, because the tasks he is working on are not related. Next the series of related tasks must have a beginning, middle and end. Building a transmitter building is a project. Operating the equipment in the building is not a project. If the tasks are developmental rather than operating then the tasks become a project.

WHAT IS PROJECT MANAGEMENT

The field of project management is divided into four areas, strategic planning, tactical planning, managing and evaluation. Each area meets a specific need in management and should be given sufficient attention in detail.

The proper way to understand strategic planning is to stand back, look at the overall picture including both the present and future perspectives. Then address these questions, "Is this a viable project?" "Does this meet the long and short term needs of my company?" What would be the impact on the company if we did not do the project? Strategic planning evaluates the needs of the company in relation to the project vs the cost to the company, their position in the market and the future needs of the company.

TACTICAL PLANNING

Tactical planning looks at the individual aspects of a project in detail, such as what tasks are needed to finish the project, who will be the one to do the individual tasks and how long will these tasks take? How much will each task cost. What is the relationship of one task to another,in time, equipment and people? Proper tactical planning requires detailed knowledge of each task and its relationship to the overall project. Computer project management is designed to automate the paper work of managing these relationships.

MANAGEMENT

To properly manage a project, we should implement the tactical planning by subdividing the various charts comprising the concept of Critical Path Method. By utilizing the information contained in the charts and from analysis of the relationships between the tasks, we can control the work with a minimum of lost time and money.

The Pert Chart

The Pert Chart (figure 1) is a visual aid comprised of small boxes containing the name of the task, the resources, the time required to do the task and the starting and ending time of the task. After each task is assigned a box with the required information the task boxes are linked in order of their occurrence. The concept of linking is the basis of project management and should be understood completely. The charts presented in figures 1,2, and 3 are a reduced version of the actual program that was used to build the tower. Notice the double dark line linking tasks 1 to 2 to 3....to 10. This is the critical path. Changes on this path will move the project completion date. The lighter line connecting tasks 5 to 6 to 10 for example is a non-critical path and minor changes may not change the critical path; however, changing the duration of a non-critical task can cause that task to move to the critical path, therefore changing the ending time forward or later. This is a useful tool in projecting "what if" cases such as bad weather, missing materials or labor problems. Major changes show up as spectacular movement among the tasks on the Pert chart. Some small modifications only change a single date and are not noticed as easily. Minor time movements are better seen in the Gantt chart.

Looking at the project of constructing antenna tower bases, the first step was to accept bids. The resource allocated for this task was assigned to the chief engineer and the time required to open, evaluate and accept a contractor was one day. When this project was planned, three parallel activities were to be carried on. The anchors for the guy cables, the actual tower base and the building to house the transmitter were to be constructed simultaneously. The resources allocated were the surveying company (Gold Engineering Co), the contractor, (Triangle Construction Co.) a drilling company (North Drilling) and the chief engineer (Born). The object of the planning session was to minimize the amount of time the various resources are on site and to assure that there are no time conflicts between the tasks assigned to a single resource.

The survey crew was to lay out the tower base and the guy anchors and later to lay out the foundation of the building. The drilling company was to dig the tower base with a back-hoe and to drill six anchor holes. The contractor was to pour the holes, the base and the building foundation, then clean the area in preparation for the tower erection. Using computer project management, each task was assigned a number, a resource, and the amount of time necessary to complete the task.

Each task was then linked to the next succeeding task. The computer then completed the Critical Path and non-critical paths for the project, thus completing the Pert Chart while checking for conflicts in resources and preparing a Gantt chart and a Resource Calendar.

The Gantt Chart

The Gantt chart (figure 2) shows the scheduled time of each task on a time line basis. Each symbol in the chart represents a day. The upper case "X" represents days on the critical path and may not be changed without effecting the completion date. The

">" greater-than symbol are float or delay days. The lower case "x" represents non-critical path tasks that can be moved or delayed.

Notice the "Form the building" (task 008) requires two days to complete represented by the symbol "xx" followed by several ">" symbols. The project manager can quickly see that he can delay the forming of the building as much as a week without delaying the project. However, Task 009 which is pouring the building foundation starts one day later than task 008 and would be delayed by the same amount. Task 009 has several days of free float available. Pouring the building foundation can be delayed up to seven days without delaying any part of the project. The Gantt charts allow a manager to see what he can do timewise to a specific task without disrupting the total project. Likewise, the manager can enter a change in a "what if" question to see what that change would do to the project completion time. Making a change, then running the program, not only changes the Gantt chart, but updates the Pert Chart and the Project Calendar. Such changes can be made in either the Pert or Gantt chart data screens for a quick "what if" look.

The Project Calendar

The project calendar (figure 3) contains all holidays that apply to the project as well as weekends. Each day is assigned a number of hours to work, usually eight; however, the number of hours can be increased or decreased as needed on a daily basis and hours over eight can be assigned as overtime hours. Rain days can be inserted into the general project calendar and a new calendar can be generated with a new general schedule.

An individual resource calendar is assigned to each resource, and all information from the general project calendar is transferred to that calendar along with the exact work days from the pert chart. (Changes in the Pert or Gantt charts will automatically be reflected into these calendars.) An example of the value of calendars is the question, should I pay overtime to complete a task before the weekend or hold the crews over and work them straight time next week? Holding a crew over the weekend involves paying expenses for lodging and food, and it delays the project. However, paying time and a half or double time could be more expensive. The question of time vs money moves the discussion to yet another chart.

Project Details

The Project Details Chart (figure 4) contains information with the billing rate for each resource, the overtime rate, and fixed expense costs for food and lodging.

With all the information for each resource entered in the calendars and project details chart, the answer to the question can be found quickly by generating a report using the overtime situation and generating a second "what if" situation using the "hold over" scenario and choosing the less expensive option. The calculations will be quicker and more accurate as the computer will catch that single task that must be completed on time when manual checking may overlook it.

EVALUATION

Evaluation is the final step in your project. Step back and look at what you have accomplished. Did you accomplish your goal? Did your resources produce and cost as expected? Were your time estimates close to the actual times? Pull out your original charts and compare with the final working charts and analyze your mistakes for future use. The evaluation stage prepares you for the next project. You gain insight into the value of services, people and equipment. Careful analysis of each person's performance reveals strong and weak areas that can be utilized, corrected or avoided as necessary. The time required for a piece of equipment to do a task should be compared to what a larger unit would cost to do the same task in a shorter time. The key to evaluation is to look for detail. Some useful information will jump out at you, other information is buried and you must dig for it.

CONCLUSION

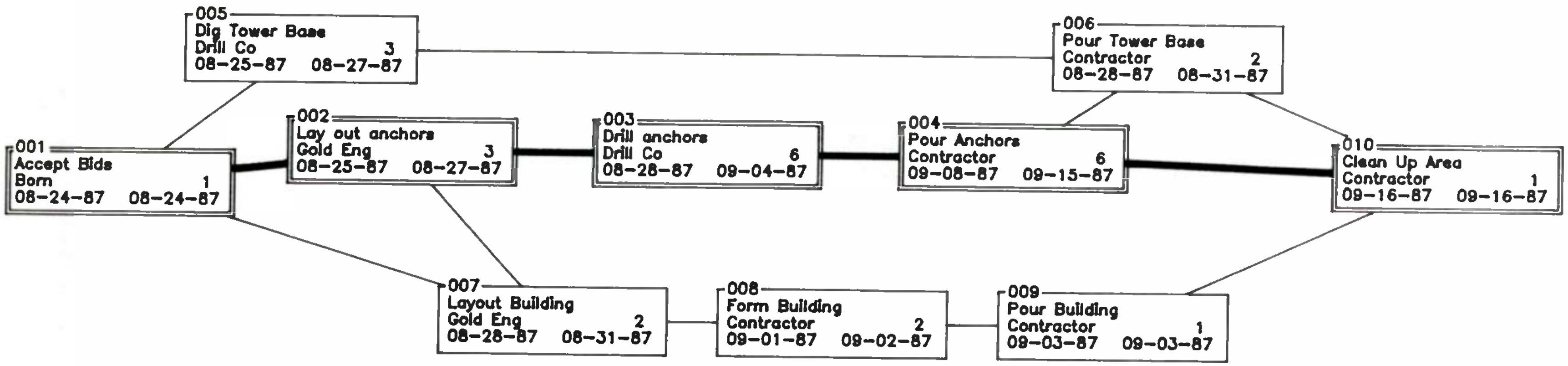
Computer project planning is a tool that allows you to plan your work in detail. While this is useful, the real value is the ability to re-evaluate and re-plan as you go along and know what each adjustment will cost in time and money. You can evaluate multiple schedule options to arrive at the optimum plan for your project.

The average broadcast project will fit into a desktop computer with room to spare. Consider how handy your friendly computer will be with project management, a CAD program, word processor and spreadsheet installed . Your whole engineering department from design to scheduling to budgets can be handled on one machine. And if you still have some time left you can read your mail box from the news department.

Acknowledgments:
Primavera Systems Inc.; Balla Cynwyd, PA
Computer Associates; San Jose, CA
Rebecca Born, MBA; Corpus Christi, TX

Tower Project KRIS-TV

Project: CPM-DEMO.PJ
08-30-87

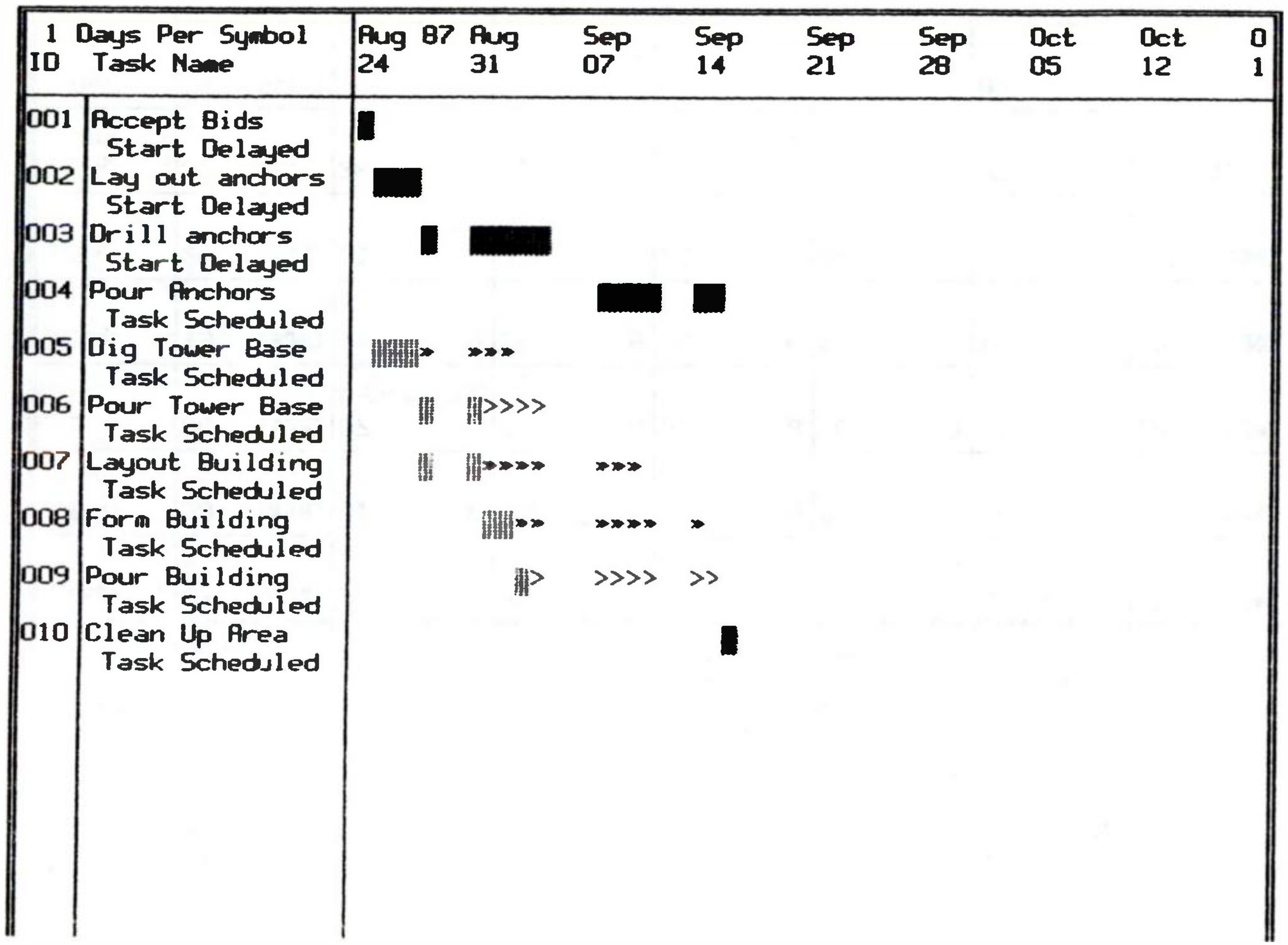


FS— SS— FF— Link — Task — Selected Task
FS═ SS═ FF═ Critical Link ═ Crit Task ═ Sel Crit Task

Task Gantt
08-30-87 23:40

Project: CPM-DEMO.PJ
Revision: 7

Figure 2



Resource Calendar
08-30-87 23:53

Project: CPM-DEMO.PJ
Revision: 7

Figure 3

Calendar for: Contractor   Number of units: 1

1987	Sun 0	Mon 8	Tue 8	Wed 8	Thu 8	Fri 8	Sat 0
Aug	23 WKND	24	25	26	27	28 8	29 WKND
Aug Sep	30 WKND	31 8	01 8	02 8	03 8	04	05 WKND
Sep	06 WKND	07 Labor Day	08 8	09 8	10 8	11 8	12 WKND
Sep	13 WKND	14 8	15 8	16 8	17	18	19 WKND
Sep	20 WKND	21	22	23	24	25	26 WKND
Sep Oct	27 WKND	28	29	30	01	02	03 WKND
Oct	04 WKND	05	06	07	08	09	10 WKND
Oct	11 WKND	12	13	14	15	16	17 WKND
Oct	18 WKND	19	20	21	22	23	24 WKND
Oct	25 WKND	26	27	28	29	30	31 WKND

Resource Details
08-31-87 00:03

Project: CPM-DEMO.PJ
Revision: 7

Figure 4

+

Rsrc Name:Contractor

		Defaults		Totals	
Costs:Begin		Hours:	40	Var:	2400.00
Total Overscheduled: 0	Rate Mult:1.50	Fixed:	0.00	Fix:	0.00
Calendar Variance: 0	No. Units: 1	Rate:	25.00	Tot:	2400.00
Sun Mon Tue Wed Thu Fri Sat		Priority: 50		Act:	0.00
Workday 0 8 8 8 8 8 0		Allocation:	8x	Hrs:	96

ID	T a s k	Pr	Hrs	Allc	Un	Ovr		Dur	Start	Finish	
006	Pour Tower Base	50	16	8x	1	0		2	08-28-87	08-31-87	
008	Form Building	50	16	8x	1	0		2	09-01-87	09-02-87	
009	Pour Building	50	8	8x	1	0		1	09-03-87	09-03-87	
004	Pour Anchors	50	48	8x	1	0		6	09-08-87	09-15-87	C
010	Clean Up Area	50	8	8x	1	0		1	09-16-87	09-16-87	C

THE ENGINEER AS AMBASSADOR: THE ROLE OF THE BROADCAST ENGINEER IN RADIO FREQUENCY INTERFERENCE RESOLUTION

by

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An article in "High Fidelity" magazine in March, 1986 entitled "RX for RF Interference" noted that some radio frequency interference (RFI) to home electronic equipment stems not from the amateur radio operator or CB'er next door but from commercial broadcast stations. The recommendation to the RFI victim was as follows:

> If your RFI is from a commercial radio station, you may get some help from its chief engineer. Some chief engineers are harassed and crotchety. Others are downright talkative and full of good ideas. It's worth a phone call to find out.

The initial response of an engineer in this instance will inevitably determine not only the extent of the problem, but public opinion of the station as well. It can be anticipated that in instances of RFI the victim of the interference will not expect to hear that the source of the interference, the "causation in fact" as the lawyers put it, is not in the radio transmitter, but in the home electronic equipment receiving the interference. It requires the utmost diplomacy on the part of the engineer in handling these complaints, lest the RFI victim "escalate" the confrontation. This article will examine the obligations of the broadcast station to resolve instances of interference, and the proper role of the engineer as the representative of the station in resolution of such problems.

I. FCC PREEMPTION OF RFI REGULATION

In cases of RF interference appearing in home electronic equipment, the station has little to fear from state and municipal courts. In 1982, Congress granted the FCC the authority to establish minimum performance standards for home electronic equipment and systems to reduce their susceptibility to interference from radio frequency energy.[1] In granting FCC this authority, Congress was careful to specify that the resolution of RFI problems was a function exclusively reserved to the FCC:

> The Conference Substitute is further intended to clarify the reservation of exclusive jurisdiction to the Federal Communications Commission over matters involving RFI. Such matters shall not be regulated by local or state law, nor shall radio transmitting apparatus be subject to local or state regulation as part of any effort to resolve an RFI complaint. The Conferees believe that radio transmitter operators should not be subject to fines, forfeitures or other liability imposed by any local or state authority as a result of interference appearing in home electronic equipment or systems. Rather, the Conferees intend that regulation of RFI phenomena shall be imposed only by the Commission.[2]

The authority granted the FCC by Congress has not been exercised, however. The Commission has stated that it prefers to rely on voluntary industry efforts to address RFI susceptibility of home electronic equipment. This is a rather curious approach,

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1 47 U.S.C. §302(a)(1982).

2 See Section 108 of the Communications Amendments Act of 1982, Pub. L 97-259, 1982 U.S. Code Cong. & Ad. News at 2277.

given that the marketplace failure to address the RFI problem was the catalyst for the 1982 enabling legislation. Nonetheless, the Commission in a Memorandum Opinion and Order released October 19, 1986 (1 FCC Rcd 289) stated the Commission's intent to "monitor the effectiveness of the efforts of the (electronic industry). If it is found that voluntary efforts are not yielding an adequate degree of RFI susceptibility control with a reasonable time frame, the Commission may then consider mandatory standards and a means of enforcing such standards." In the intervening five years since the Commission was granted authority to establish RFI susceptibility standards, the number of RFI complaints received at the Commission has not decreased to any significant degree. It remains to be seen what a "reasonable time frame" is for voluntary industry standards to be implemented.

Though the Commission has not acted to require the improvement of RFI susceptibility in home electronic equipment, it has faithfully protected its licensees against local lawsuits filed by RFI victims, and attempted regulation of RFI by local governments. The Commission has consistently claimed exclusive jurisdiction over RFI. For example, in a Memorandum Opinion and Declaratory Ruling, FCC 85-578, released November 4, 1985, the Commission invalidated a requirement contained in a conditional use permit issued by a Klamath Falls, Oregon, zoning authority that an FM facility must protect co-located TV translators from interference.

As part of an effort to upgrade the facilities of KJSN(FM), Klamath Falls, the licensee, 960 Radio, Inc. filed a modification application with the Commission to relocate the KJSN antenna to Stukel Mountain. In its FCC application, 960 Radio stated that it would take steps to eliminate, if possible, any interference to pre-existing facilities on Stukel Mountain. These included TV and FM translators, two-way DPLMR, and FAA facilities. The Klamath Falls zoning board, however, conditioned the grant of zoning approval for the new tower on non-interference. 960 Radio sought a preemption ruling from the FCC. In granting the request, FCC stated as follows:

> We observe that exclusive jurisdiction to resolve questions involving interference has been assigned to the FCC. See, e.g., 47 U.S.C. §§ 152(a) 301, 303(c), (d), (e) and especially (f). While it has never been held that federal legislation in the field of broadcasting excludes every possible application of state law to radio stations, the Supreme Court has stated that the FCC's jurisdiction "over technical matters" associated with the transmission of broadcast signals "is clearly exclusive." Head v. New Mexico Board of Examiners in Optometry, 374 U.S. 424, 430 n. 6 (1963), See also National Broadcasting Co. v. United States, 319 U.S. 190, 217 (1943) (the Commission has "comprehensive powers to promote and realize the vast potentialities of radio"); and Federal Communications Commission v. Pottsville Broadcasting Co., 309 U.S. 134, 137 (1940) (the Commission is to design a "unified and comprehensive regulatory system for the industry").

State courts as well, with the exception of those of New Jersey, have recognized FCC's preemption of RFI. In Helm v. Louisville Two Way Radio Corporation, 667 S.W. 2d 691 (Ky. 1984), the Kentucky Supreme Court held that even though the entire Jefferson County Police Department radio system was rendered repeatedly inoperable by interference from a mobile telephone facility, a nuisance claim by the police department could not be sustained:

> The relief asked by Helm to bring this fact situation within the jurisdiction of the state court on the theory of common law nuisance would necessarily involve prohibiting or controlling radio transmissions. This regulatory power has been preempted by Congress in a situation such as we have here, and the common law nuisance remedy is not sufficient to give the state court jurisdiction. Helm's remedy is with the Federal Communications Commission and not in the courts of this Commonwealth.

As mentioned above, only New Jersey seems reluctant to accept the concept of FCC preemption of RFI. Repeatedly cited in New Jersey courts, is the case of Bynum v. Winslow Township, 181 N.J. Super. 2 (App. Div. 1981) ("there is no express or implied intent by Congress to exercise exclusive control over the actual operation of amateur radio transmission . . . Our courts have recognized that local government retains regulatory interests outside of those areas controlled by federal regulation"). Though the Communications Amendments Act of 1982 clearly overruled this case as a matter of fact, several townships in New Jersey continue to maintain RFI ordinances which hold the transmitter operator strictly liable for instances of RFI, regardless of technical fault. The FCC General Counsel's Office has on several occasions been willing to offer opinions in specific cases relative to the FCC's preemption of RFI, which have served to convince municipal attorneys of the lack of state and local jurisdiction.

II. FCC EXPECTIONS OF THE BROADCASTER

Complaints of RFI each year, received by FCC, number in the high 60,000 range. In Fiscal Year 1985, the number of complaints attributed to Part 73 licensees was 2,746. The number of interference complaints obviously outstrips the ability of the Commission's Field Operations Bureau to resolve, and they rely on voluntary cooperation from both the transmitter operator and the receiver of RFI, regardless of technical fault.

Broadcast engineers are generally aware of the FCC's rules regarding blanketing interference contained in Sections 73.88 (for AM stations) and 73.318 (for FM stations). The obligation of AM stations to "satisfy" all "reasonable" complaints of blanketing interference within the 1 volt-per-meter-contour, and the obligation of FM stations to resolve complaints of blanketing interference within the 115 dBu (562 mV/m) contour not related to mistuned receivers, improperly installed antenna systems, or use of gain antennas or antenna-mounted booster amplifiers, are relatively straightforward.

Interference problems not attributable to blanketing interference, however, are not addressed by FCC rules. To the extent that the problem is technically attributable to the susceptibility of home electronic equipment of the RFI recipient, and thus resolvable only there, the assistance and good-faith efforts of the engineer are called for, albeit with the caveat that the engineer should not accept responsibility for resolving the problem. In some cases, especially those involving expensive hi-fi amplifiers and tuners, the recipient of interference might best be referred to the contact representative of the manufacturer of the susceptible device. The FCC's Interference Handbook, 1986 Edition, which is available either from the FCC Field Offices or from the Government Printing Office, Superintendent of Documents, Washington, D.C. 20402, lists contact representatives of most major electronic equipment manufacturers. Referral of the RFI complainant to the manufacturer and offering a copy of the Interference Handbook may resolve many minor problems. Chapter 3 of that book addresses FM Transmitter interference, and generally refers the complainant to the FM station's chief engineer, except in cases of interference channel 6. In the latter cases, the complainant is referred to the television station. The Handbook suggests that if the complainant is not satisfied with the engineer's response, to contact the nearest Commission Field Office.

Jack Brooks, CE at HF Broadcast Station KGEI, Redwood City, California, has produced a collection of typical RFI problems and solutions which may assist the engineering community in resolving complaints directly at the complainant's residence or business. A compendium of RFI materials is also available through EIMAC in San Francisco. The station engineer might well prepare a procedure manual for his station to use.

If the station engineer is non-responsive in an interference case, and yet the interference is attributable to the susceptibility of the offended device, then the FCC may ask the station to demonstrate that a similar type device (such as a different brand of telephone) does not exhibit the impaired operation in the same RF environment, thus establishing where the "technical fault" lies. This is used both as a means of avoiding direct FCC involvement and to convince the complainant that indeed the problem is with his own equipment.

III. THE STATION'S AUDIENCE AND RFI

Interference from RF generated by the engineer's station is far less frequently encountered, however, than is interference to the station's own listeners from other sources. Interference to AM listeners, for example, from RF lighting devices, power line leakage, and other sources of interference again require that the station engineer, as a matter of his/her own self-interest, be available to solve certain RFI problems. For an AM station, inevitably the most insidious problem is power-line leakage, which is almost never reported to FCC (or to the station) by listeners. The NAB, in the face of FCC statements that interference to broadcast reception is "not a problem", conducted a poll which revealed that 60% of the respondents had heard static interference to AM broadcast stations. The alert AM station engineer will establish a rapport with the local power company radio office, and occasionally check through the station's coverage area for serious power line leakage. The Commission will probably not be much help in compelling the power company to keep lines clean (despite Section 15.25 of the FCC

Rules) in the absence of a large number of listener complaints. That rule section requires that the operator of an incidental radiation device resolve all interference.

Aside from power line noise, most receiver interference problems stem from devices in the home, or from other broadcast stations. While interference from the former is not often reported, interference from other broadcast stations arouses the listener's ire. The Commission's policy in cases of RFI from a new station is clear:

> Whether by imposition of specific conditions or by operation of law, a licensee building a new facility is obligated to take all the necessary steps, including the financial burden, to correct interference problems caused by new or modified construction. See e.g. Sudbrink Broadcasting of Georgia, 65 F.C.C. 2d 691 (1977); Athens Broadcasting Co., 68 F.C.C. 2d 920 (1978); Midnight Sun Broadcasting Co., 11 F.C.C. 119 (1947); B & W Truck Service, 15 F.C.C. 2d 769 (1986).

Recently, FCC reminded Part 22 DPVMRS licensees of their obligations in this regard, relative to tower installation within 2 miles of an AM directional array, or within 1/2 mile of a non-directional array.

In Minnesota, individual members of the WCCO-FM and KSJN-FM listening audiences sued numerous other stations, claiming that the defendant's stations "distorted" the transmissions they wished to receive. FCC had investigated, and had required defendant to reduce power from 100 kW to 50 kW. Further FCC testing was not sufficient to resolve the problem, so plaintiffs sued in state court, alleging that defendant's transmissions were a nuisance. The Supreme Court of Minnesota affirmed the dismissal of that suit, as FCC had preempted all state regulation of RFI. Blackburn v. Doubleday Broadcasting Co., 353 N.W. 2d 550 (Minn. 1984). The case arose prior to FCC's promulgation of its blanketing interference rules. Prompt FCC enforcement might avoid a similar situation today. However, the case points up the need for the station's engineer to be "in touch" with the station's audience relative to technical problems and reception problems, both as a means of protecting the station's image in cases of interference to home electronic equipment from the station's transmitter and by protecting its own audience from external sources of RFI.

NRSC PROGRESS REPORT

By Michael C. Rau

National Association of Broadcasters
Washington, D.C.

In June, 1985, NAB and the Electronics Industries Association (EIA) reactivated the National Radio Systems Committee (NRSC) as a joint effort of the broadcast and receiver manufacturing industries to focus on the technical problems of AM broadcasting. (The NRSC was initially formed in 1980 to conduct tests of FM receivers for submission to the FCC). The NRSC mechanism is simply a forum where broadcast and AM receiver engineers can meet to productively discuss mutual issues of concern. In January, 1987, the NRSC released a voluntary national standard for AM preemphasis, deemphasis and bandlimiting. But the NRSC has continued to work on related industry issues. This paper describes this work and the NRSC's agenda for 1988 and beyond.

Introduction: The NRSC Audio Standard

After eighteen months of development, in January, 1987, the NRSC released an "interim" voluntary national standard covering AM transmission preemphasis (figure 1), AM receiver deemphasis (figure 2) and bandwidth of the audio signal fed to the AM transmitters (figure 3). This standard has been submitted to the American National Standards Institute (ANSI) for formal voluntary standardization.

There are many reasons to install this standard in an AM station. Some benefits include (1) more effective use of your AM channel, (2) reduction of intermodulation distortion, and, of course, (3) a significant reduction in "splatter interference" that today adversely affects second-adjacent channel AM stations. As of September 28, NAB has received 304 postcards (in all sized markets) confirming conversion to the NRSC standard. A survey of audio processor manufacturers, however, shows that nearly 700 NRSC kits have been sold. Not a bad start for the NRSC's first standard.

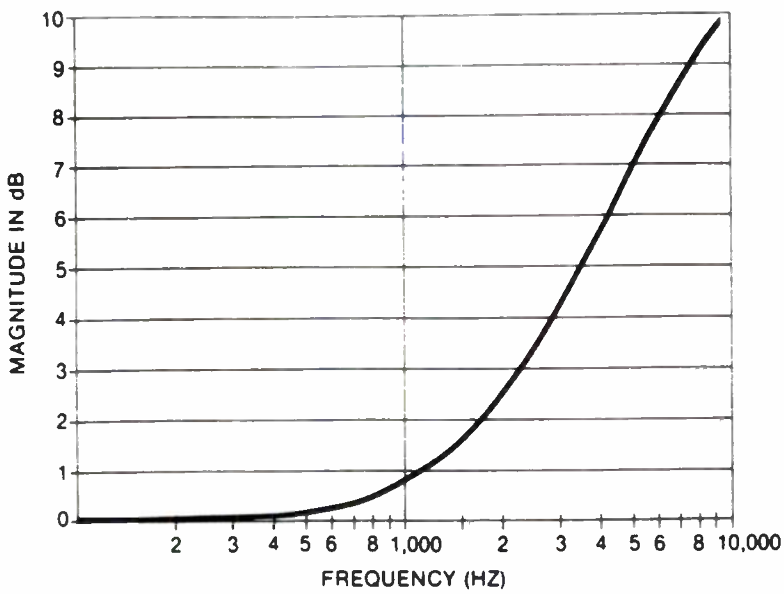

Figure 1. Modified 75μs AM Standard Preemphasis Curve

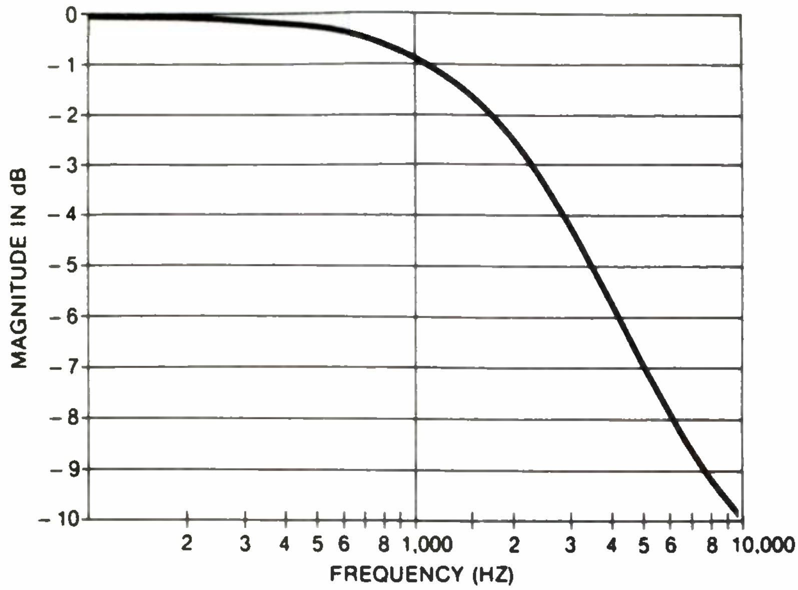

Figure 2. Modified 75µs AM Standard Deemphasis Curve

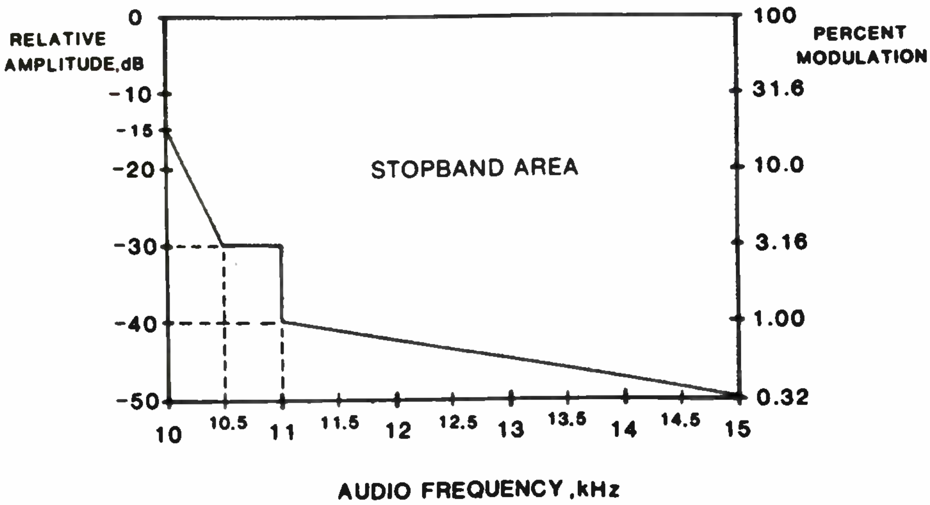

Figure 3. NRSC Stopband Specification
(Audio Envelope Input Spectrum To AM Transmitter)

The reduction in intermodulation (IM) distortion is particularly significant. Figure 4 shows the results of a standard IM test on a plate modulated transmitter. A pair of tones is fed to the transmitter, and IM distortion is measured at the output of a modulation monitor. Note that IM distortion increases with increasing tone pairs. Limiting bandwidth to 10 kHz will lower your stations IM distortion. Similar tests were performed on a PDM and a Doherty type modulation scheme, with similar results. These tests probably understate the IM distortion due to their steady-state nature. Under dynamic program conditions, there is the added source of IM distortion caused by audio signals that slew faster than the RF circuits can accurately reproduce. Moreover, any impedance mismatch between the transmitter and antenna system will cause some power to be reflected back, mixing in the transmitter's plate circuit. Implementing the NRSC standard should immediately help the quality of your signal.

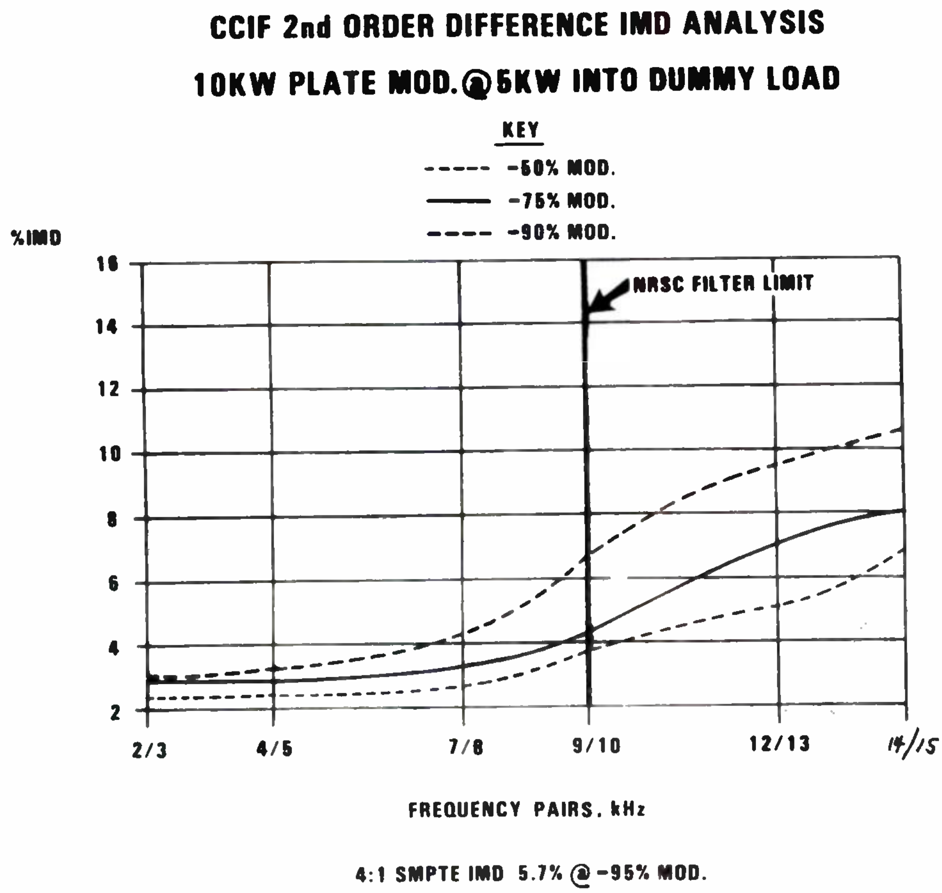

Figure 4. Intermodulation Distortion and the NRSC Audio Standard

RF Mask Development: 1987 and 1988

In September, 1987, the NRSC released a draft voluntary standard "RF Mask." This document describes the RF complement to the (audio) NRSC standard. It is a proposed emission limitation for AM broadcasts. The mask accounts for the modulation and transmission process. A six month public comment period was established to afford all interested parties an opportunity to support or oppose the proposal (deadline: March 11, 1988). Both the audio and RF NRSC standards include measurement procedures and standardized test signals.

The NRSC's proposal for an RF Mask is actually two standards proposals. The first is designed to be measured "off the air" with a conventional sweep-frequency spectrum analyzer. Figures 5 and 6 (dotted line) describe this standards proposal. It is designed to be measured when the AM station modulates ordinary program material conforming to the NRSC audio preemphasis and bandwidth standard.

The second proposed standard is a "test and control standard." See figures 5 and 6 (solid line). This standard should aid AM transmitter troubleshooting by providing AM a repeatable test that measures bandwidth occupancy.

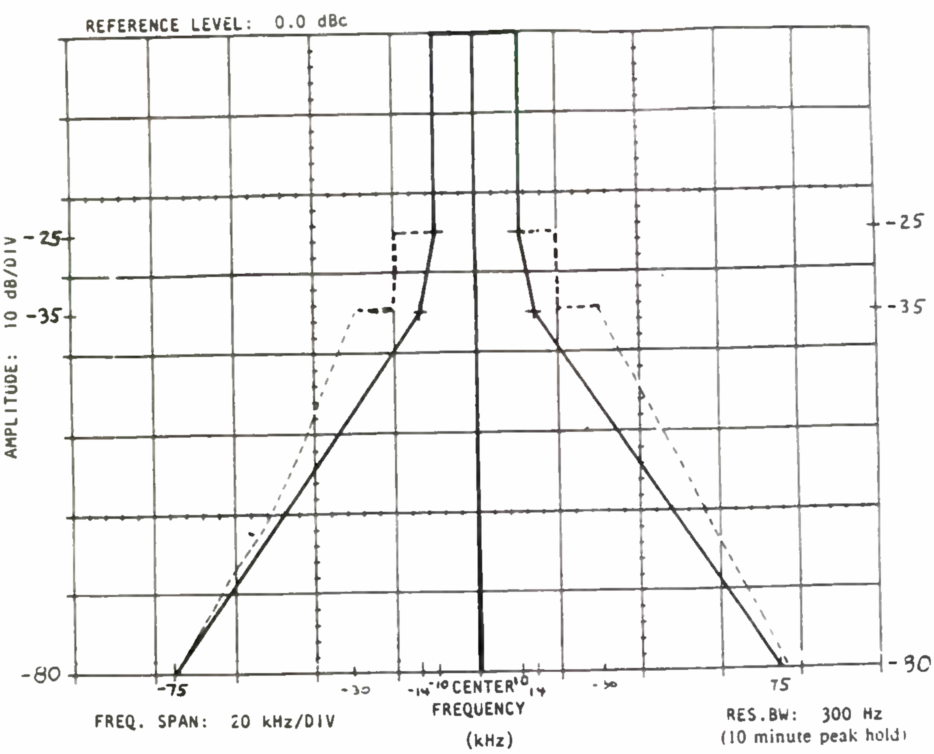

Figure 5. National Radio Systems Committee Draft Emission Limitations for AM Broadcasting

- - - - Maximum Occupied Bandwidth
______ Testing and Control Standard

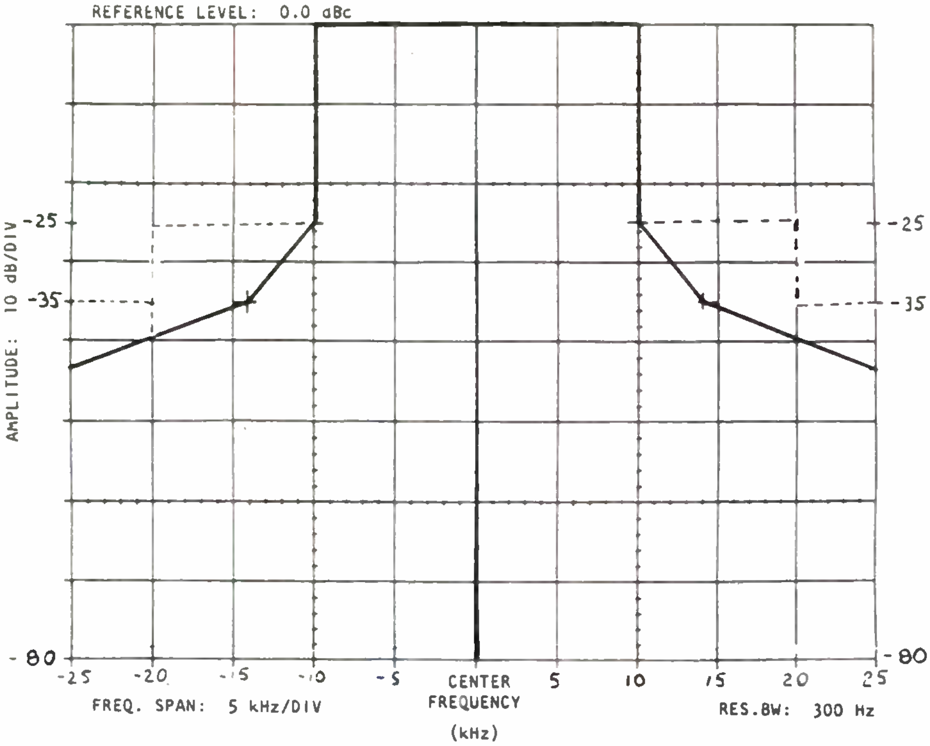

Figure 6. (Expanded scale). National Radio Systems Committee Draft Emission Limitations for AM Broadcasting

- - - - Maximum Occupied Bandwidth
______ Testing and Control Standard

But the most interesting development to arise from the NRSC's RF Mask effort is the concept of a "splatter monitor." This device may one day replace the ubiquitous AM modulation monitor. The whole purpose behind FCC rules regarding overmodulation and applicable measurement devices is to prevent interference to other AM stations. A 1986 NAB technical report examined AM overmodulation and showed that the current rules do not address a given AM station's potential for causing interference. Among other findings, the report showed that interference may just as easily be caused by properly modulating a clipped audio signal as by driving unclipped audio past the point of carrier pinch-off. Not exceeding 100% negative peaks, therefore, does not guarantee a "clean" spectrum, especially if the audio is being clipped prior to modulation.

A splatter monitor would enable a station to observe interference being caused to adjacent channels, regardless of the transmission mechanism that produced the interference. What is important in determining an absence of interference is how a consumer receiver will behave when listening to stations on adjacent channels. Accordingly, the splatter monitor is little more than an AM stereo receiver with meters. The monitor must detect both amplitude and phase information occurring on adjacent channels. It would be limited to a 10 kHz bandwidth and include NRSC deemphasis. The detected audio output would be "filtered" by the CBS loudness algorithms to correct the audio for a listener's subjective reaction to interference (what is important is not the absolute magnitude of the splatter monitor's readings, but a correlation of its readings with the objectionability of interference). The NRSC is still considering this "splatter monitor." It may be that such devices will one day be available on a commercial basis.

FCC Notice of Inquiry: 1988 and Beyond

The FCC has released a " Notice of Inquiry" proceeding that leaves open the question of the entire set of technical standards our current AM allocation system is based upon. It is a large document that addresses nearly all the AM technical and allocations issues there are to address. The NRSC presently is working on a report for submission to the FCC. Comments on the Notice are due December 17; however, there is a some chance that this date can be extended if the industry needs more time to complete its work. Among the technical issues in the Notice are (1) the appropriate value of minimum usable field strength; (2) co-channel protection ratios; (3) first and second adjacent channel protection rations; (4) groundwave and skywave propagation algorithms; (5) calculation methods, in particular how to handle multiple interfering signals at the protected contour; and (6) appropriate emission limitations on AM broadcasts. In addition, a variety of non-technical issues are raised as well.

It is too early to predict the outcome of this Notice. However, the report of the NRSC should be a very comprehensive document and will be available in April, hopefully at NAB's Spring engineering conference. Its focus is on values of technical standards that are necessary to promote the development of high-fidelity AM broadcasting. It will not deal with AM stereo issues, only allocations standards. But what should the industry and FCC do with new standards? NAB hopes that new standards can be developed and implemented in the expanded AM band, 1605-1705 kHz. For the existing band, it is not clear how the new standards could be put to their best use without massive reallocation of stations. Before these issues can be considered, we need to do the industry's technical homework. These are issues for 1989 and 1990. Presumably, we all would like to see a successful AM industry for the next several hundred years.

Conclusion: Some Random Thoughts

These are not easy projects and there are no easy answers. So far, the NRSC has succeeded in developing consensus on the kinds of difficult choices that are capable of effecting a major industry improvement. Support the NRSC. No other industry-wide engineering committee is working on achieving a consensus resolution of AM's difficult technical issues.

Remember that YOU *are* the AM broadcast industry.

Michael C. Rau
Washington, D.C.
September, 1987

ADVANCED TV SERVICE
THE NEXT GENERATION OF TV SERVICE

DR. THOMAS P. STANLEY, CHIEF ENGINEER
and VICTOR TAWIL

OFFICE OF ENGINEERING AND TECHNOLOGY
FEDERAL COMMUNICATIONS COMMISSION

Our topic today is advanced television. I intend to speak about two recent actions approved by the Commission relating to the developments and uses of advanced television technologies by off-air television broadcasting: The first is the Notice of Inquiry on Advanced Television. The second is the establishment of a joint FCC-Industry Advisory Committee to monitor developments in ATV. I will also highlight some of the technical areas that require industry participation and concensus so as to assist the Commission in bringing about the optimal level and form for advanced television, and insure a smooth and orderly transition to the next generation of TV service.

The topic of advanced television is not new to the Commission. In the early 1970's, the Commission conducted a series of ad-hoc studies to explore the technology of future television distribution and assess the impact of new and emerging technologies -such as satellite broadcasting and high resolution television- on the function, structure and economics of the existing terrestrial broadcast industry. In 1974, the Office of the Chief Engineer published a three volume report entitled: "Technological Boundaries of Television" that addressed advances and expectations of television technology for the next decade and made recommendations to the Commission in this area. More recently, the subject of advanced television systems has been peripherally mentioned in various Commission proceedings such as the satellite scrambling proceeding, the LM/TV sharing proceeding and the technical standards for cable station proceeding. While the subject of advanced television has periodically surfaced during the past two decades, the Commission has not initiated a formal inquiry into advanced television technologies and what role the FCC should play in their growth until earlier this year. All-in-all, I believe that the Commission's inquiry into advanced television is timely, and should be viewed as a necessary first step toward bringing the next generation of broadcast television technologies to the Nation's broadcast and video consumers.

NOTICE OF INQUIRY ON ADVANCED TELEVISION (ATV)

In July, 1987, the Commission initiated a wide-ranging inquiry into the uses of advanced television technologies by off-air television broadcasting. The objective of the inquiry is to consider the technical and public policy issues surrounding the use of advanced television technologies by television broadcast licensees. The purpose of this inquiry is to acquire information that will help the Commission better understand the advantages and disadvantages, cost and benefits of the various terrestial broadcast ATV implementation options. Having secured that information, the Commission will then be in a position to decide whether adoption of some form of advanced broadcast service would be in the public interest, and, if so, what form the system should take.

Generally speaking, the NOI inquired about the following areas:

a) The expected development and value of off-air terrestrial advanced television in the United States;

b) The features, capabilities and development status of advanced television transmission and reception systems available now or under development or whose development is foreseen, and the extent to which these new technologies can be used by other video media; such as DBS, MMDS/ITFS, cable, VCRs and video discs;

c) The allocation and technical issues that need to be investigated and resolved prior to implementation of advanced television systems for television broadcasting;

d) The economic, legal and regulatory issues that need to be addressed and resolved prior to implementation of advanced television systems:

e) The perceived public interest implications for the existing television broadcast service in the United States;

And finally

f) The general timetable for implementation of advanced television systems in broadcasting.

Realizing that there are numerous ways of significantly improving the technical quality of television, the Commission elected to define for the purpose of this proceeding the term "advanced television system" in a very broad and generic sense so as to encourage the widest participation possible. Specifically, the Commission considers any system(s) that improves television audio and video quality or enhances in any way the current NTSC system as an "advanced television system". This definition encompasses all systems now under development or whose development are foreseen, including -but not limited to- to what is commonly described in the industry as improved NTSC systems, enhanced 525-line systems, and high definition systems.

I. ATV Technologies.

In the Notice, the Commission noted that development efforts for improving audio and video quality have taken many forms, and are constantly being challenged by expanding technologies. This is evident by the number and variety of proposals that have been developed over the years and the ones that are currently under investigation. While the present range of ATV technologies is broad, covering a great number of approaches and development strategies, most of the development work relating to ATV technologies concentrated in one or all of three areas: a) video/audio quality performance, b) compression of transmission bandwidth, and c) compatibility with the present NTSC system. In particular, development work focused on better understanding the interrelation between these three areas and the various trade-offs associated with transmission of improved TV images. Among the tradeoffs that are under active investigation are: quality-for-bandwidth tradeoffs, quality-for-compatibility tradeoffs, increased receiver cost for reduced transmitted bandwidth tradeoffs, etc.

In the inquiry, the Commission is seeking information on questions such as

1) How quickly are developments of the various ATV technologies progressing? Which are now operational? Which are in prototype stage? Developmental stage? How long until these systems are realized?

Also,

2) What changes in ATV technologies should be anticipated for the near future? Can ATV technologies be expected to develop so that the transmission bandwidth of a high resolution production source can be compressed to fit within 6 MHz channel without apparent loss of quality?

And

3) What criteria should the Commission use to evaluate and compare the various ATV technologies? Are Criteria, such as the

video/audio quality performance, transmission bandwidth, NTSC compatibility, etc. appropriate? What are the appropriate trade-offs between the various criteria?

II. Anticipated Nature of ATV

Implementation of an advanced television service could be provided one of three ways: One) as a new service separate and distinct from the existing broadcast service, similar to the current AM and FM broadcast services, Two) as a service that augments wherever feasible the existing NTSC service with no provision for full replacement of the NTSC service, similar to the current direct broadcast satellite service where the DBS service complements the terrestrial service, OR, Three) as a service integrated fully with the existing broadcast service which over time will eventually replace entirely the existing NTSC service similar, to the integration of FM stereo within the FM service. Depending upon the expected demand and value of off-air terrestrial advanced television to the nation's television viewers, it is conceivable that one implementation scenario would be more desirable than another.

As a related matter, the Commission is also interested in determining the desired technical features for this new service. For example, whether stations which propose to transmit advanced television systems should be modelled on the same technical planning considerations used for the existing broadcast service, such as a single transmitter per area, a 50- to 60-miles service contours, indoor and outdoor receiving installations, or whether the Commission should select different technical planning considerations, such as smaller service contour, multiple transmitters to cover a larger service area and outdoor receiving installations only.

III. Spectrum Options

By far the most important issue the Commission is addressing in the ATV proceeding, is whether additional spectrum would be necessary to implement ATV, and where it will come from. As you probably are aware, there are a number of ATV transmission systems currently under development that require spectrum capacity greater than assigned or available to TV broadcasters. While it is premature for the Commission to express its views at this stage of the proceeding regarding whether additional spectrum would be necessary to implement ATV, it is wise for the Commission to explore the various alternatives for providing additional capacity for ATV systems in the event it concludes that providing additional spectrum capacity for ATV is in the public interest.

Basically, the Commission described three possible options for providing additional spectrum capacity for ATV. The first option entails the use of the existing VHF and UHF television allocations. The second, the sharing of spectrum with other broadcast or non-broadcast services or the use of reallocated or unallocated spectrum. The third, is a combination of the previous two options.

As you probably are aware, it may be possible to implement some kind of an ATV service within the existing VHF and UHF spectrum. Additional spectrum capacity could, for example, be obtained through the adjustment or elimination of the current broadcast-to-broadcast interference protection standards, such as the cochannel, adjacent channel, or the UHF taboos, or through partial or total "repacking" of the VHF and UHF bands. While these approaches are feasible, the Commission has not fully investigated them to determine the amount of spectrum that would be made available and how it may be used for ATV. Basically, the Commission is interested in determining whether it should implement an ATV at VHF or UHF or both, if so, how much spectrum would be made available, and what would be the technical and economic impact on the existing NTSC service if the Commission modified and/or eliminated the existing protection criteria to accommodate ATV, or attempted to partially or totally rearrange the existing channel assignments.

It may also be possible to implement an ATV service by allocating presently vacant frequencies or by sharing frequencies now allocated to other services. Spectrum below 1 GHz, while best suited for terrestrial broadcasting, is already extensively used by broadcast and other non-broadcast services, so that any reallocation of existing non-broadcast spectrum could adversely impact existing operations in that portion of the spectrum. Above 1 GHz, ATV could possibly share spectrum with the existing ITFS/MMDS services at 2.5 GHz, the direct broadcast satellite service at 12 GHz or any of the bands that are allocated for point-to-point purposes. Here again, while these spectrum possibilities are feasible, they have not been fully investigated to determine their suitability and desirability for terrestrial broadcasting or the impact on the non-broadcast services. Essentially, the Commission is interested in determining whether it should implement ATV on non-broadcast spectrum, and if so, in what portion of the spectrum and how much. What are the technical and economic advantages and disadvantages of this spectrum option compared to the previous one, and what is the impact of sharing non-broadcast spectrum on ATV to the non-broadcast services.

An ATV service might also be accommodated using a combination of the two options I just described. For example, advanced TV stations could be authorized on both the existing VHF and UHF television bands and, when necessary, in new spectrum. This option becomes especially attractive if it is determined that the amount of additional spectrum capacity available in the existing VHF and UHF television bands is not sufficient to provide for the complete replacement of the NTSC service.

Based on my current knowledge of advanced television systems, I am inclined to say that, if it is determined that additional spectrum is necessary, the most promising and regulatory expeditious approach for the implementation of an advanced television broadcast service is to utilize frequencies already allocated for television broadcast use. Such an approach would encourage existing broadcasters to participate in the implementation of such a service, would foster compatibility with the existing NTSC service, and most of all would promote more intensive use of the existing broadcast spectrum.

IV. Compatibility

Another important issue the Commission is addressing in the ATV proceeding, is "compatibility". Compatibility is a term that is generally used by the various proponents of ATV technologies to describe their system in relation to the current NTSC system. The term has often been the subject of many debates with no general consensus on what it should describe. For example, some have used the term compatibility to describe the relationship between their transmission system and the NTSC standard, while others have use it to describe whether their system could be received by a conventional NTSC receiver or whether it could be operated within a 6 MHz channel. To eliminate this confusion, we will apply the term "compatibility" in a specific sense so as to establish consistent guidelines for evaluation and comparison of the various ATV technologies. Specifically, an ATV system is considered to be compatible with the existing television service if it operates within the present 6 MHz channelization scheme and if the ATV signal can be decoded and viewed on a conventional NTSC receiver. The Commission is especially interested in determining how well the various ATV technologies fit this definition of compatibility, and if incompatible, what would be required to make these systems compatible. Also, what are the consequences if the Commission requires compatibility, and what are the advantages and disadvantages to the broadcasters, equipment manufacturers and the nation's broadcast viewers, if the Commission relegates the compatibility issue to market place forces.

Generally speaking, in considering the evolution to advanced broadcast television, I am inclined to say that compatibility is an essential ingredient needed to insure a smooth and orderly transition to advanced television, especially if the implementation of ATV is in the existing VHF and UHF broadcast spectrum. The importance of compatibility, however, varies depending upon the various implementation scenarios and spectrum options envisioned for ATV. For example, under some implementation scenarios, such as allocation of new non-broadcast spectrum to ATV, compatibility with the existing NTSC service would not be essential but rather a desirable feature. All-in-all, it appears -to me at least- that the degree of decisional significance one attaches to compatibility depends to a large extent on the spectrum selection and implementation scenario envisioned for ATV.

V. Relaxation of NTSC Standard and UHF Taboos

Implementation of advanced TV systems may, of course, require adjustments to many of the the Commission's technical requirements for television broadcast stations. For example, the existing NTSC transmission standard would probably have to be relaxed to allow the use of improved transmission standards. Additionally, If additional spectrum capacity will have to be obtained from the existing UHF broadcast band, relaxation of the mileage separation requirements between alloted UHF channels could allow more efficient usage of the UHF spectrum for advanced television. With that in mind, the Commission is specially interested in determining whether it is in the public interest to continue to require the mandatory NTSC transmission standard, or whether it is more beneficial to make the standard voluntary in light of the fact that relaxation of the standard or some aspect of it could facilitate the introduction of some ATV systems. Furthermore, the Commission is interested in knowing the extent to which the UHF taboos continue to be necessary for the protection of existing service, and what effects the taboos may have on the implementation of advanced television systems.

Finally, I would like to mention that the Commission's Inquiry addresses many other important issues that I will not have the opportunity to touch upon today. I, therefore, urge each one of you to acquaint yourself with these issues, and actively participate in the proceeding so as to assist the Commission in satisfactorily resolving these issues.

JOINT FCC-INDUSTRY ADVISORY COMMITTEE

The other Commission action I intend to address today is the joint FCC-Industry Advisory Committee on Advanced Television. The committee has been established to advise the Commission on the facts and circumstances regarding advanced television systems. The committee's tasks are to assemble and analyze information dealing with technical, economic, legal and regulatory issues, deliberate upon appropriate policies and actions, and ultimately develop recommendations regarding the planning, design and implementation of ATV in the U.S.

In establishing the Committee, the Commission was particularly interested in creating a structure that provides for a balanced representation of appropriate points of view without creating an unwieldy, administratively cumbersome entity. The result is a two-tier organizational structure. The higher tier is the parent Committee, the second tier is the working subcommittees. The parent Committee is a "blue ribbon" committee composed of senior industry officials and nationally recognized expects in the field of television broadcasting, including broadcasters, programmers, manufacturers, and academicians. The committee is expected to provide the Commission with momentous advice on policies and activities regarding advanced television as well as to review and endorse specific advice from the working subcommittees. The Commission expects the parent committee to reach a consensus on most, if not all, of the issues it deliberates upon and report its recommendations as expeditiously as practical.

As to the working subcommittees, the Commission established three separate subcommittees: A planning subcommittee, a system subcommittee, and an implementation subcommittee. Membership on these subcommittees is open to representatives from all interested organizations.

The planning subcommittee is responsible for determining the attributes of an advanced television service. Specifically, the subcommittee is tasked with defining the desirable characteristics for the new service, in terms of picture quality, population served, cost to broadcasters/manufacturers/consumers, relationship to existing broadcast services, etc., and with recommending planning factors for this new service such as coverage area, quality of service, frequency reuse criteria, receiver interference immunities, etc.

The system subcommittee is responsible for specifying the appropriate transmission and reception characteristics for providing advanced television service in the United States. Specifically, the subcommittee is tasked with evaluating and ultimately selecting, on a technical and economic basis, advanced television system(s) under development for possible implementation in the U.S. In addition, the subcommittee is also tasked with determining the appropriate transmission/reception standards and spectrum requirements for the recommended system(s).

The implementation subcommittee is responsible for establishing a scheme for the implementation of ATV in the U.S. Specifically, the subcommittee is tasked with developing a transition plan to ATV including the appropriate policies and regulations to oversee the implementation.

Generally speaking, the Commission envisions that the majority of the work will be performed by these three subcommittees, since they will formulate the recommendations regarding the specific desirable features of ATV, develop the appropriate systems facilities and standards, and establish plans for implementation of ATV. In essence, I forsee that these three subcommittees will play a very active and decisional role in the overall process. I therefore encourage each one of you to actively participate in one or more of these subcommittees.

The Commission expects the advisory committee to address the important issues of spectrum capacity and compatibility very expeditiously and to submit an initial written report on spectrum requirements and the fundamental parameters for ATV by the spring of next year.

I would like to conclude this presentation by highlighting some of the technical areas that I believe require industry participation and consensus in order to bring about the optimal level and form of ATV. One area is the identification and agreement on a single set of technical system parameters for the purpose of properly comparing the performance and spectral efficiency of various candidate ATV systems. Another, is the development of a universally accepted methodology or procedure for evaluating these various systems. Currently, there is no general consensus in the industry on how to compare and rank the video/audio quality performance of these various ATV systems relative to each other, or on a single accepted methodology to determine their relative spectral efficiency. I believe that it is imperative that the television industry actively take-on these areas and work toward coming up witn a consensus methodology for evaluating the various ATV systems.

Spectrum Management Database

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Editor's Note:

The following report was prepared by the author and presented to a spectrum management conference in Geneva, Switzerland, at the request of the FCC.

United States of America

Interim Working Party 1/2

A SAMPLE SPECTRUM MANAGEMENT DATABASE SYSTEM USING "OFF THE SHELF" DATABASE MANAGEMENT SOFTWARE

1. Introduction

One of the key areas of concern in spectrum management is the exchange of accurate and timely data among users. A database has been developed using off the shelf Data Base Management software, for the more than 100 Broadcast Auxiliary Service (BAS) Frequency Coordinators in the United States who are recognized by the Society of Broadcast Engineers (SBE). The BAS uses radio spectrum in various bands both mobile and fixed to facilitate or enhance Television and Radio broadcast operations. Typical examples of this use include relay of program material from studio to transmitter site, a mobile station relaying programming from a news event, hand-held and mobile units used for voice communications directing the programming material in the field, and relay of telemetry data from transmitter site to studios. These frequency assignments cover from 26.0 MHz to 19.7 GHz, and in some cases the frequencies are shared with other non-broadcast licensees, both fixed and mobile. For the most part, fixed operations are primary with mobile operations secondary in the 1.9 GHz (1,990-2,110 MHz), 6.8 GHz (6,875-7,125 MHz), and 12.7 GHz (12.7-13.20 GHz) fixed and mobile bands. This sharing results in special problems since news events covered with mobile frequencies can not be predicted and the problem of coordination with fixed and other mobile users has to take place on a "real time" basis. These coordinators work on voluntary frequency coordination efforts in their local areas to help see that interference between users is minimized. The work of these coordinators is similar to the work done on a national level by many administrations except that the local frequency coordinator has to work on "short time" notice to facilitate information exchange among users of BAS spectrum. The database program was developed to establish a standard data format for exchange of information throughout the United States.

2. Background Information

2.1. History

Since the middle 1970's, and the great increase in Electronic News Gathering (ENG) in the United States, it has been apparent that improved spectrum management techniques are needed. Thus it became the goal of the broadcasters to develop an accurate listing of all users of the BAS bands, and basic technical parameters of operations.

The initial database for the Dallas, Texas, area consisted of a list kept in a word processing system. Updates of the list were made available every 6 months. The most basic data was included -- frequency, station call sign, power output, receiver location, and antenna azimuth. Perceiving a need for more information, and recognizing that soon there would be more users than frequencies identified using such a simple database system, a program for a micro-computer was developed to store data. The system required 2 disk drives for storage and was extremely slow in processing. It required extensive knowledge of data entry techniques and procedures. The system, although slow, allowed interactive query from remote locations. This micro-computer system was unsatisfactory for general use by coordinators, whose experience with computers varied widely and which was rapidly improving.

Next, work began developing a data base using a sophisticated "data base" management programming language. By early 1984, the programs were written which allowed entry and retrieval of data concerning the frequency usage during the news broadcasts from the 1984 United States political conventions in Dallas and San Francisco. The system was further enhanced when it was made available "On-Line" by telephone modem on a 24 hour basis beginning 4 months before the conventions, through the final night of those broadcasts. Many users took the opportunity to call the system and make inquiries, or post new information about their own operations. A duplicate system was on site for "real time" coordination during these special events.

2.1. Current Design Goals

There are a number of goals inherent in the current SBE database program. These goals were recognized from practical limitations in present hardware and software designs:

1. Low or no cost acquisition of the program without "USER/LICENSE FEES".

2. Widespread availability through many sources including bulletin board systems, etc.

3. User friendly operation, a minimum of operator intervention.

4. Data integrity.

5. Ability to exchange database files among users in various regions by electronic means or mailed floppy disks.

6. Ability to allow a user to form a national database, if that user wanted to obtain database files from other areas and combine them.

7. Flexibility for future expansion as changes arise, and through user experience.

3. Technical Information (as of the year 1987)

3.1. Hardware Requirements

3.1.1. Minimum Hardware

IBM-PC or Compatible Computer System
2 - 360K Floppy Disks
1 - 720K Floppy Disk
Monochrome or Color Monitor
256K Memory
Hayes Command Compatible 1200 Baud Modem

System limitations will allow storage of approximately 750 records on a single 360K floppy disk, or the program itself and 1000 records on a 720K floppy disk.

3.1.2. Preferred configuration

IBM-AT or Compatible Computer System
1 - 1.2 Megabyte Floppy Disk
1 - 20 Megabyte Hard Disk
Monochrome or Color Monitor
640K Memory
Hayes Command Compatible 2400 Baud Modem
Dot Matrix Printer

This system will accommodate storage of well over 6,000 records with the needed index keys for retrieval. Information can be backed up to a floppy disk for archival retrieval. Possibility to use an expanded memory machine as host of Local Area Network System for additional users.

3.1.3. Ultimate Configuration

COMPAQ 386 or Equivalent 80386/20 MHz Equipped Computer
1 - 1.2 Megabyte Floppy Disk
1 - 130 Megabyte Hard Disk
1 - Streaming Tape Backup System
EGA Color Monitor
Hayes Command Compatible 9600 Baud Modem
Laser Printer

This system would easily accommodate storage of over 60,000 records, with speed matched only by mainframe-style computer systems. This system, as the server on a Local Area Network, could support many simultaneous users without major speed-of-retrieval degradation.

3.2. Software Requirements

Only the MS-DOS operating system or equivalent is needed to operate the SBE frequency coordination database program. The program is compiled into an executable form, and produces DBASE(tm) compatible data files, memory files and report forms. The key index files (.NTX) files are unique to CLIPPER(tm), and can not be used directly by DBASE(tm) products. Future versions of

CLIPPER(tm) are scheduled to use the more familiar (.NDX) index files of DBASE(tm).

4. Design Goal Achievements

4.1. User friendly operation, with a minimum of operator intervention.

The CLIPPER(tm) compiler has enhanced menu operations which are used extensively within the program. All database functions are "MENU BAR" driven. This means that each selection on the menus is highlighted through the use of cursor up/down arrows, or by pressing the first letter of the prompt word. The "ESC" key can be used at any time to abort the operation in progress, and return to the last menu or prompt. Continued pressing of "ESC" will eventually return to the Main Menu. An orderly exit of the program can take place, using the "QUIT" selection. When entering data, many times one will have similar information to enter. Data entered is carried over to the next new entry, but by pressing "HOME", old data is erased, a time saver for repetitive data entry. During data entry, the actual database is not updated until an affirmative response is received to save the data. This insures that a possible power failure during entry will not corrupt data files.

4.2. Data integrity

While searching the database by frequency or station call sign, only the database administrator is allowed to edit data already on file or delete records from the database. Appropriate prompts are used to assure that the record is, in fact, intended to be changed or deleted. To further enhance the deletion, the record is not physically removed from the database until the files are reorganized; an operation which should be done from time to time to eliminate unwanted records, or after problems arise. When changing data already in the database, the actual data is not changed or updated until the final prompt asking for confirmation of the changes. After the affirmative answer to this prompt, the information is written back to disk. This ensures that changes were intended to be made. The accuracy of any data contained within the database is dependent on the source of the data, it's age, and it's verifiability by other users. Additional information obtained by the database administrator from various sources can be of great value. Every record in the database also has a date on which the last information update of any kind took place.

4.3. Ability to exchange database files among users in various regions.

(Ability to allow a user to compile a national database, if that user wanted to obtain database files from other areas and combine them.)

Within the format of the database layout (Appendix A), are fields which allow data taken from various geographical areas, to be combined into a larger database. The delays in updating make a database only as accurate as it's most recent entry. A National database for SBE purposes, would be un-needed from many standpoints, but the ability to use one database program with many areas databases would be helpful for network and other itinerant users. The data base fields; local, state, city code and area code make date exchange possible. For example, all data entered in Dallas, Texas carries the following state, city and area codes: "TX,001,214". Further, information which is contained in the database which is not directly within the Dallas coordination area is marked with field "LOCAL" as F or False. Data from Austin, Texas is easily included in the Dallas data base, by changing the state, city and area code fields to "TX,002,512". Internally the program is able to recognize that multiple city data has been entered and allow searches with areas filtered out as needed.

4.4. Future enhancements as changes come about through user experience.

No system for frequency coordination could be written once, then never changed again. Continued user input is expected and necessary. This could be considered just the beginning of a more complex system, which can make a frequency coordinator's job easier. The database entry and retrieval system is the first of a two-part system to make information available. The establishment of a "Frequency Coordination Network", using an electronic bulletin board system (BBS), will provide unattended information exchange. In studying the many bulletin board systems which can be obtained, one in particular is very user friendly, easy to operate, and lends itself well to information exchange. It might include for example: message handling capability, both public and non-public with the ability to send these messages to other BBS systems. These messages to and from other BBS may have database files or information requests attached to them. File uploading and downloading using numerous computer protocols is enhanced.

5. Summary

Although intended for a very specialized application by the Society of Broadcast Engineers, the database program was created in one engineer's spare time. Countless hours have been spent making the system more user friendly and easier to operate for the many persons who will come in contact with the system.

APPENDIX A:

1. The FREQDB data base:

This is the main file which will be the heart of a standardized database system for all coordinators in the country. The following layout is made to be expanded if nessecary by the committees. As each new item is added to this list, it will be incorporated into the national database standard. By following this layout, in this order, further additions can be accommodated very easily.

Field	Field Name	Type	Width	Dec	Description
1	REFNUM	Numeric	10		Used by the Coordinator to identify this system/relate to DETAIL File
2	FREQUENCY	Numeric	10	4	Frequency to 4 Decimal Places Use Center Frequency for Video
3	STATION	Character	10		Station Call Sign or Company Name
4	SYSCALL	Character	8		Part 74 Call Sign or /APPLICATION/PENDING/PROPOSED
5	RECVCITY	Character	20		City where receiver is located
6	USE	Character	30		What this system is used for like News Dispatch, STL--Vertical, ICR
7	HOWMUCH	Character	40		A description of how often or frequent this frequency is used.
8	REMARKS	Character	55		Special Information or data which would be helpful, Repeater In/Out Frequency
9	AZIMUTH	Character	10		Azimuth of Main Lobe for transmitters, deg.
10	UPDATED	Character	8		Date of Last update for this information
11	CONTACT	Character	20		Primary Contact person at this station
12	PHONE	Character	14		Telephone Number (XXX)-XXX-XXXX for this Station - Not a private number
13	LATITUDE	Character	10		Transmitter Site Latitude (DD-MM-SS)
14	LONGITUDE	Character	10		Transmitter Site Longitude (DDD-MM-SS)
15	RLATT	Character	10		Receiver Site Latitude (DD-MM-SS)
16	RLONG	Character	10		Receiver Site Longitude (DDD-MM-SS)
17	TANTENNA	Character	20		Transmitter Antenna Type/Description
18	RANTENNA	Character	20		Receiver Antenna Type/Description
19	POWER	Numeric	5	1	Power out in watts to Antenna
20	EIRP	Numeric	8	2	Equivalent Isotropically Radiated Power, dBW
21	TLOC	Character	55		Physical Address of Transmitter Site
22	RLOC	Character	55		Physical Address of Receiver Site
23	EMISSION	Character	10		Emission Designator (16KOF3E, 25MOF8F)
24	GROUND	Numeric	6	1	Transmitter Antenna Ground Level Elevation AMSL, feet
25	COR	Numeric	6	1	Transmitter Antenna Height AMSL at Center of Radiation , feet
26	PERISCOPE	Logical	1		Does this System use a Periscope Antenna Y or N (T or F)?
27	REFLECTOR	Logical	1		Does this System use a Reflector?
28	PASSVREPTR	Logical	1		Does this System use a Passive Repeater?
29	LOCAL	Logical	1		Is this system a LOCAL user inside your normal area of Coordination ?
30	STATE	Character	2		State Abbreviation
31	CITYCODE	Numeric	3		3 Digit Code to be assigned by NFCC
32	AREACODE	Numeric	3		3 Digit Code to be assigned by NFCC
33	ID	Logical	1		Does this system use Vertical Blanking Interval or some other type of ID'er?
34	IT	Logical	1		Could this Frequency be shared with Proper Coordination with Itinerant Users?
35	NU	Numeric	3		Number of Units which are used on this frequency (An indication of Heavy Usage)

2. The DETAIL data base:

The next file called DETAIL will be related to the FREQDB file by REFNUM. The detail file, which will not be available initially by Remote Access, will be for detailed information about the transmit and receive antennas. This should assist areas which need to do what has been termed as "3 dimensional coordination". This information would have to be obtained from the company putting the system on the air.

Field	Field Name	Type	Width	Dec	
1	REFNUM	Numeric	10		Used to Link this Database with FREQDB
2	TMFG	Character	20		Transmitter Antenna Manufacturer
3	TTYPE	Character	10		Transmitter Antenna Type
4	TDBI	Numeric	4	2	Transmitter Antenna Power Gain in dBi
6	T3DB	Numeric	3		Transmitter Antenna 3 dB Point
7	RMFG	Character	20		Receiver Antenna Manufacturer
8	RTYPE	Character	10		Receiver Antenna Type
9	RDBI	Numeric	4	2	Receiver Antenna Power Gain dBi
11	R3DB	Numeric	3		Receiver Antenna 3 dB Point

3. The STATION data base:

The third file, which will be related to FREQDB by STATION, will contain the detailed information about the station contacts such as Master Control Phone Numbers, Private Lines, Pager Numbers, Home Phone Numbers, Transmitter Sites etc. This file will not be available to the general public, only to members of the Coordinating Committee and those that they wish to have the information. The format for this file has not been determined, we are looking at various options.

NATIONAL CONVENTION

BROADCAST
ENGINEERING
CONFERENCE

Proceedings
of the 1987 SBE & Broadcast Engineering Conference

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