THE SOUND ENGINEERING MAGAZINE

db

SEPTEMBER 1980
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Jim is one of the good ol' boys of Nashville. His engineering career stretches back some 18 years to the days of mono mixing. He's done everything from pop to R&B to disco—and, of course, country. The aviation industry gave Jim his technical background. But he's also prepared himself by playing four or five different instruments. Some of the names on the other side of the glass from him include Bob Dylan; Simon and Garfunkel; Peter, Paul and Mary; Loretta Lynn; Johnny Cash; Don Williams; Marty Robbins; Conway Twitty; Ray Price; and Roy Clark.

ON SPECIALISTS

"Let me say that I have sympathy for them, because they're missing the rest of the world of music. They're locked into one thing and I got it all. I have done four different styles of music in one day. I did a disco record that got to number six on the Billboard charts, 'Dance With You.' In the same day, I did a number one country record. You don't listen to the same kind of music all the time. And I don't want to listen to the same kind of music all the time, either."

ON OVERPRODUCTION

"'Swarm.' That's my term for over-production. I've had producers who have turned and said,'Well, how many tracks have we got left?' You may look at the chart and say, 'Well, we've got nine tracks left.' He'll say, 'Great.' And he looks into the window of the studio. 'Hey, let's put an electric piano on.' Not because the electric piano fits the song and has a place or meaning in the rhythm or in the feel of the song, but it's because he sees one in the room and we've got nine tracks to go. And that's overproduction, abuse of multitrack recording. And that I don't condone."

ON PLAYBACKS

"I actually mix. I don't load tape. I like to sit down at the console, set my monitor levels equal and put the band together and get a monitor mix in the control room that sounds as close as I can make it to the record, so that the producer and the artist and the musicians can hear and understand what they're doing and correct their mistakes. I'm an old mono mixer. And that's what built mono mixing."

ON TAPE

"A competitor of 3M has stated that 3M has a greater print-through than their product. It's my opinion that there is no greater print-through on the Scotch® 250. It's just not masked with modulation noise. There also was a comment that the competitor's tape was brighter, when in fact, there was just more third harmonic distortion in the 10 to 12 kc range. I am very stringent on monitoring in the control room. And when I hear a signal off the floor, I want it to come back off the tape the same way. I don't want it to be embellished with third harmonic distortion to make it brighter, or modulation noise to confuse the bass line."

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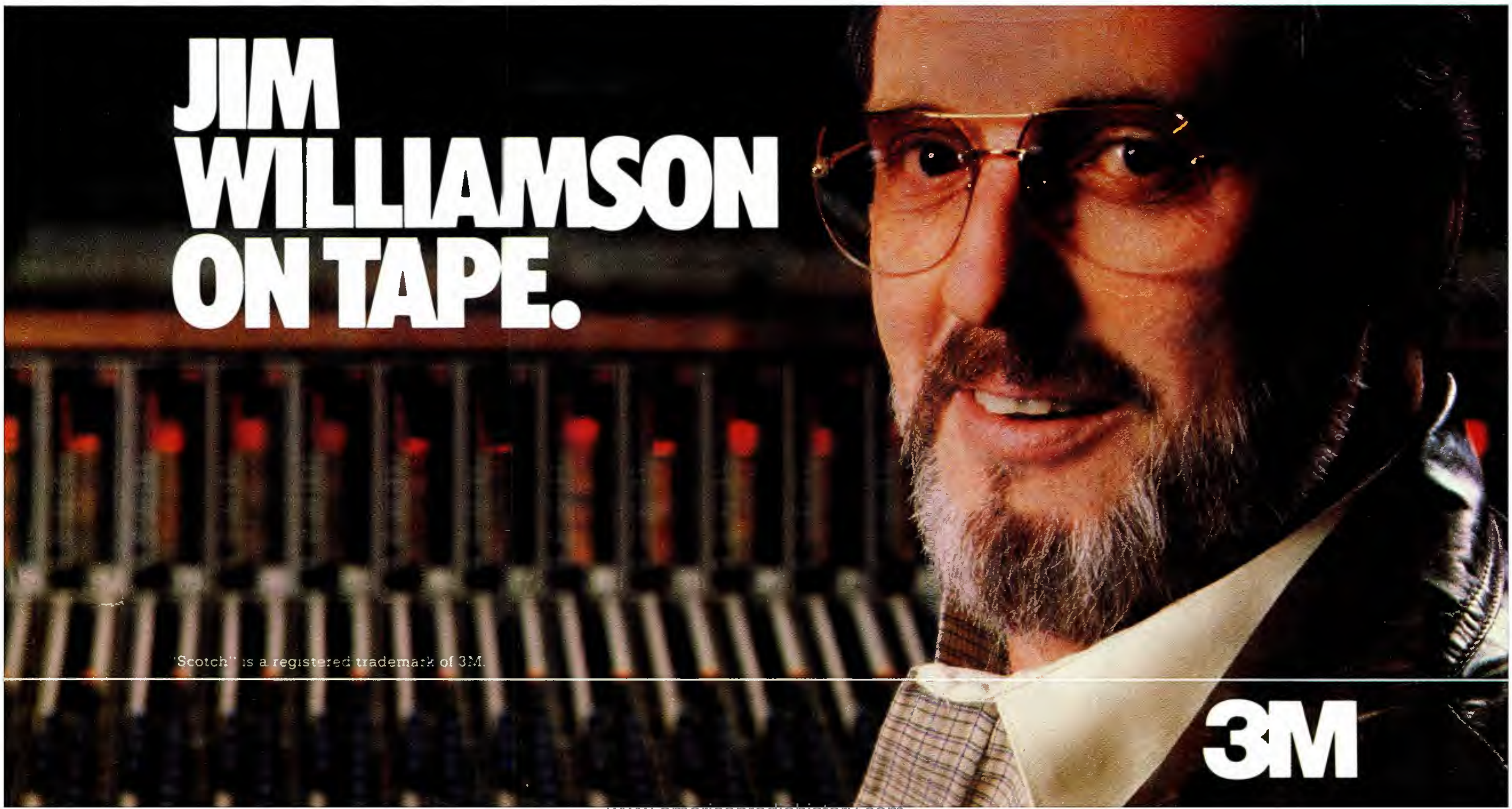

Coming Next Month

- Is your studio really humming (at 60 or 120 Hz)? Are you infinitely baffled by standing waves? Were you "off the air" for an hour, and no one complained?
- You may need help, and next month we've got some, as our authors tackle the ways and means of getting and keeping your studio's system and acoustics in good shape.

THE SOUND ENGINEERING MAGAZINE

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db is listed in **Current Contents: Engineering and Technology**

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About The Cover

- An Audiotronics 26-input Model 110A Audio Console highlights this sound effects master control facility. Installed earlier this year at ABC's Broadcast Center on 67th Street in New York, this control room serves two adjacent live effects and layover studios, as well as providing effects to twelve TV studios and post-production facilities.

Circle 45 on Reader Service Card

db Letters

To The Editor:

In John Borwick's article highlighting British professional equipment manufacturers (June, 1980), the author listed B&W Loudspeakers, but failed to name the US representative. The representative for B&W Loudspeakers in the United States is Anglo American Audio, 1080 Bellamy Road North, Scarborough, Ontario, Canada M1H 1H2.

M. J. Remington

To The Editor:

We have been reading the letters regarding broadcast audio processing with great interest. Here at WMBM we have taken every step possible to insure that all dynamic range is eliminated.

All music is taped thru two very fast peak limiters hooked up in series. The compression ratio is 200 to 1, release time is 20ms. At this stage we add 15 to 20 dB of boost above 5kHz. This is to insure optimum tape saturation. Phono cartridge tracking force has been optimized at 0.2 grams; this results in a very punchy sound to the highs. Our music is so hot it sounds like the stylus is going to jump out of the groove.

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Listener reaction? Just amazing. WMBM has replaced over 150 speaker cones damaged by DC modulation. Lawsuits are pending on 12 auto crashes resulting from listener fatigue. Cume

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ratings have jumped 350% but quarter-hour ratings have dropped unexpectedly. Here at WMBM we are proud to have solved the audio processing dilemma.

GREG STRICKLAND
Chief Engineer

db replies:

Well done! But couldn't you raise your auto crash ratings a bit? Also, you could avoid cone damage by soldering a ten-penny nail across each listener's terminals. That would really take care of your quarter-hour problem as well (No charge for this advice.)

TO THE EDITOR:

Before running out to buy a reel of two-inch videotape, expecting 5 MHz response from your 24-track, consider a few facts of recording life.

No direct analog magnetic recording system is good for reproducing substantially more than a range of ten octaves. While any properly set up professional video machine will record and reproduce a bandwidth exceeding 5 MHz, (greater than eighteeen octaves) demanded by a high resolution color video signal, this is accomplished by frequency modulating a "carrier" between 7 and 10 MHz, (less than one octave). The head-to-tape writing speed required to handle this frequency range is 1500 ips, accomplished via a rotating headwheel carrying four heads. Complex capstan and head servosystems along with various timebase error reduction circuits help to restore the original signal.

Now to the tape. The particle orientation for videotape used in this manner is 90 degrees referenced to the tape edge. This works well for the scanning video head, but tends to destroy the linearity of any audio tracks recorded the "regular way." The thin coatings, while firmly attached to the backing, do not lend themselves to the signal and bias levels to which audio engineers have become accustomed. This is of no consequence in video, since the FM video signal is recorded at a level approaching saturation.

Helical systems are a slightly different matter. Despite Mr. Weinstock's reference to "Helical particle orientation," SMPTE standards dictate longitudinal alignment—like audiotape—for professional-format helical-scan videotapes.

While helical systems offer greatly improved audio performance when compared to older VTR audio, the backing thickness and coating thickness again limit the audio signal drives and bias conditions to something less than those considered optimum in an audio application. Reference fluxivity for C-format audio is 100nanoWebers/Meter—nearly 8 dB below the level of a "normal" 250 nWb/M audio machine.

While I have disagreed with some of Mr. Weinstock's technical comments, I

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strongly agree with him that we are crossing into a new period of radio awareness in the field of television, to the benefit of all of us.

PALMER S. PATTISON
Audio Engineer
KUED-TV

db replies:

Oops! Change the offending sentence so that it describes superior results (when recording audio in videotape), unless particle orientation has not *been optimized for helical-scanning. This restores Weinstock's—and the tape's—original orientation.*

TO THE EDITOR:

I really appreciate the highly technical articles in your magazine. It's amazing what those of us involved in sound engineering can learn from just one periodical. Scott Pelking's article in the July issue of **db** on how to build a "PRASP" was just too good to be true. My only complaint is that Mr. Pelking didn't say what brand of undershirt works best for applying stain. However, I checked with several engineers and the consensus is that Hanes is the best, with Fruit-Of-The-Loom a close second. Perhaps you could get Scott to do some more writing for **db.** I, for one, would enjoy an article about building bookends or an end-table. Maybe he could even discuss whether it is better to use a ¾-inch or ⅞-inch hole when construcing a wren house.

ORIN N. FRIESEN
Chief Engineer
KEYN AM & FM
Long-Pride Broadcasting, Inc.

db replies:

We've always thought that gentlemen prefer Hanes, although for AC/DC operation, you may want to try Fruit-of-the-Loom.

For bookends, why not try a miniature PRASP? Actually, you'll need two of them; one for each side of the book (or books). You can also lay a PRASP on its side, if you need an end-table.

As for the birds, that's a little bit beyond our scope. However, we imagine you should measure the wrens before drilling. But, as a chief engineer, you probably know all about birdies already.

TO THE EDITOR:

We would like to take this opportunity to thank you for the article you did on Indian Creek Recording Studio in your June issue. There was however one fact (misprint) which was not correct and is very important to us and even more so to our clients. Our rates are $100 per hour instead of the $200 quoted. We feel that this is an extemely competitive rate for a studio of this quality.

JOHN ROLLO
Chief Engineer

Digital Audio

Up to this point in our discussion, we have considered the encoding process where the audio signal passes through the anti-aliasing low-pass filter, the sampler and the analog-to-digital converter. The result is a series of digital numbers. But now, let's consider the decoding process, in which this series of numbers is converted back into a continuous audio signal. These encoding and decoding operations are mirror images; each element in the encoder has its counterpart in the decoder.

Just as the analog-to-digital converter (ADC) transforms a single voltage value into a digital word, the digital-to-analog converter (DAC) transforms the digital word into a single voltage. Conceptually, the digital-to-analog conversion is much simpler, since each digital number corresponds exactly to a particular voltage. In analog-to-digital conversions, there is a quantization error, since a range of voltages within the quantization interval are all transformed to a single digital number. There is no corresponding quantization error in digital-to-analog conversions.

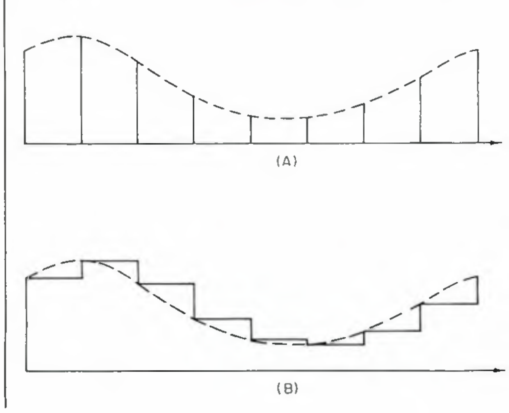
Consider a 10-bit digital word which is to be converted to an analog voltage. The 10-bit word specifies one of 1024 different voltages. Thus, the number 1 might correspond to 0.001 volts, the number 2 (0010) to 0.002 volts, the number 3 (0011) to 0.0003 volts, etc. Since there are 1024 unique voltages, this DAC would have a range of +0.511 to -0.512 in steps of 0.001 volts. (The range is asymmetric since 0.0 volts is one of the 1024 voltages.)

DE-SAMPLING

We must now consider the process by which the individual samples of the DAC can be smoothed to create a continuous audio sample. This is the reverse of the sampling process in the encoder. At this point in our discussion we need to digress, since the way of analyzing the smoothing processes is different from the way they are actually built. We may think of the DAC's output as being a series of samples, as shown in FIGURE 1A or, we may think of it as being a constant voltage between samples, as in FIGURE 1B. In both figures, the DAC produces a voltage corresponding to the digital word. However, in the top figure, the DAC produces its output for a very short "sampling" time. In the bottom figure, the DAC "holds" its voltage until the next digital word enters to change the value. Now, forget FIGURE 1B for the time being and we will return to it later on.

Since we will be using a low-pass filter to smooth the DAC's output, it would be wise to consider the spectrum of the signal represented in FIGURE 1A. By understanding the spectral content, we can see clearly how the low-pass filter creates the desired signal. FIGURE 2 shows the way in which the waveform of FIGURE 1A could have been created. FIGURE 2A shows a 1 kHz audio signal. This will be multiplied by the sampling signal in FIGURE 2B to create the result in FIGURE 2C. The sampling signal in FIGURE 2B is a periodic pulse train with a

Figure 1. (A) The sine wave is sampled at regular intervals; (B) each sample voltage is held until the next sample is taken.



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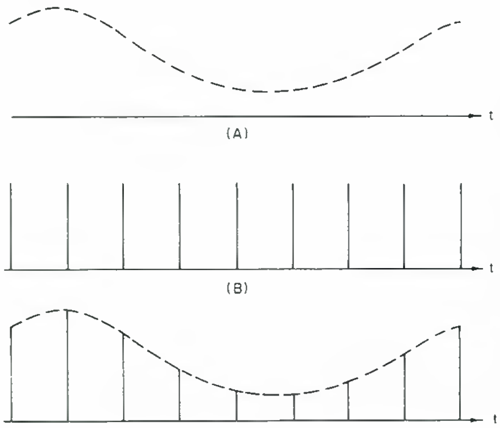


Figure 2. (A) A 1 kHz audio signal; (B) The sampling signal; (C) A × B = C.

fundamental at the sampling frequency and many overtones or higher harmonics. If the fundamental is 40 kHz, we would expect to find energy at 80 kHz, 120 kHz, 160 kHz, etc. Multiplication is like a very severe nonlinearity which creates large amounts of sidebands. This is similar to amplitude modulation or the intermodulation of two sine waves. As with intermodulation distortion, the resulting signal contains frequencies at the sum and difference of the components. The waveform in FIGURE 2C contains energy at 1 kHz (audio), 39 kHz (sampling frequency – audio frequency), 41 kHz (sampling frequency + audio frequency), 79 kHz (2 × sampling frequency – audio frequency), 81 kHz (2 × sampling frequency + audio frequency), etc.

SOUND WITHOUT IMAGES

The role of the smoothing filter is to remove all components other than the 1 kHz audio. These other components are called "images" since the spectra centered at 40 and 80 kHz are the same as the audio spectrum centered at DC. If the audio signal contains components of 100 Hz, 200 Hz and 800 Hz, then the sampled signal will also contain these components, as well as image components at 40 kHz, 40.1 kHz, 40.2 kHz and 40.8 kHz. There will also be mirror images at 39.9 kHz, 39.8 kHz and 39.2 kHz. Since the smoothing filter must remove these images, it is called an anti-image filter.

The anti-aliasing filter at the input limits the energy in the audio to less than 20 kHz in this example. But, a 19 kHz audio signal produces images at 21 kHz [40 – 19], 59 kHz [40 + 19], 61 kHz [2 × 40 – 19] and 99 kHz [2 × 40 + 19]. The desired signal at 19 kHz must be separated from the nearby undesired image at 21 kHz. For this reason, it is usually better to limit the anti-aliasing filter to a frequency lower than half the sampling frequency. This provides a guard-band for the transition betwen desired and undesired signals. An 18 kHz limit allows a 4 kHz guard band, since the nearest image is now (40 – 18) 22 kHz.

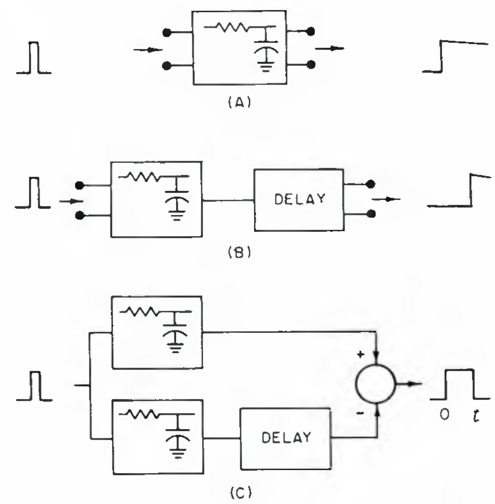


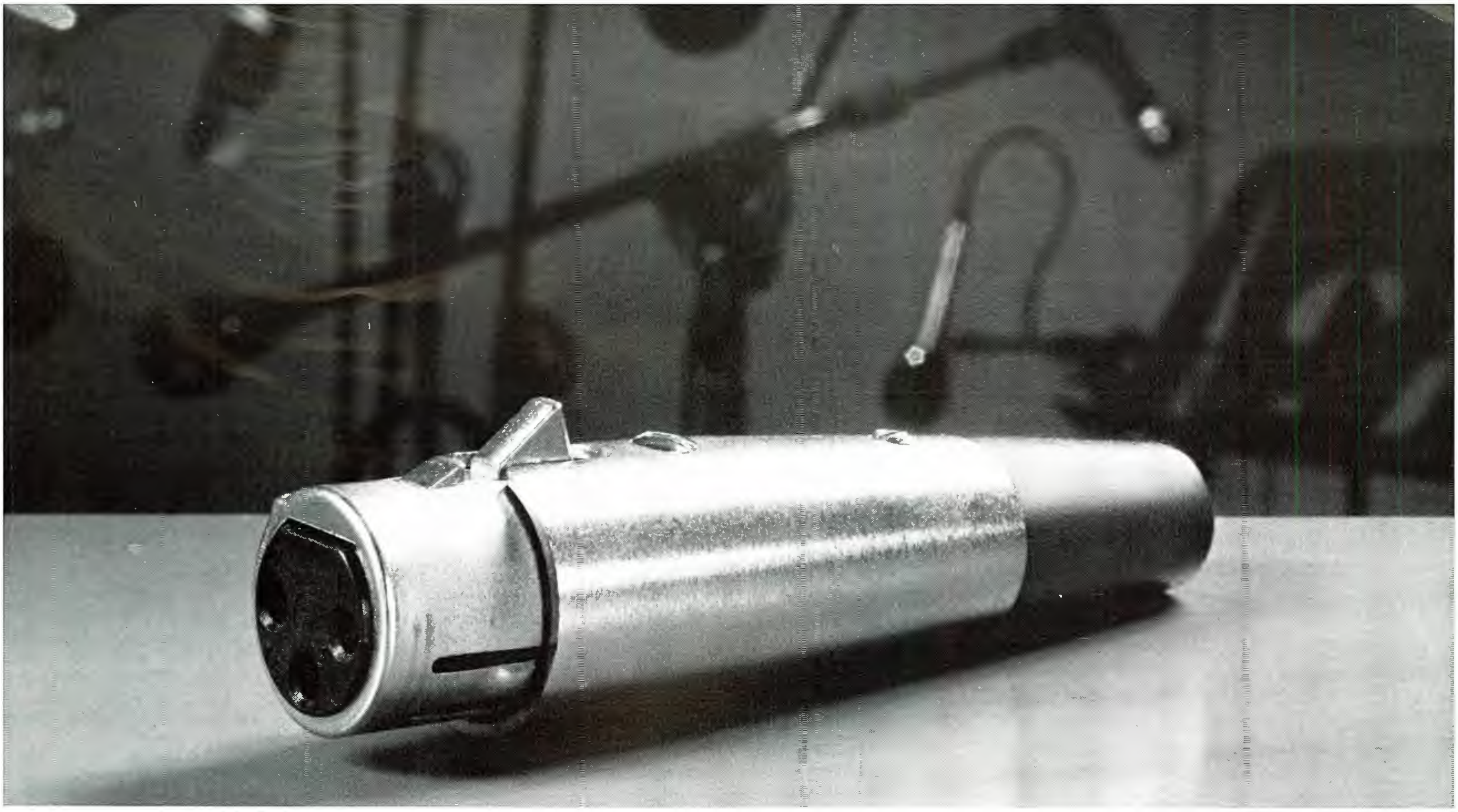
Figure 3. (A) Input and output of a low-pass filter with a very long time constant; (B) A low-pass filter with an extra delay inserted; (C) A – B = C.

SAMPLE-HOLD

We are now in a position to return to the actual implementation of the DAC using an implied hold. FIGURE 1B can be thought of as holding the sample between successive words. We must now show that the effective holding action, which turns a very short pulse into a longer rectangular pulse, is nothing more than a filter. FIGURE 3A shows the input and output for a low-pass filter with a very long time constant, which is called an integrator. FIGURE 3B shows another low-pass filter with an extra delay inserted. If these two outputs are subtracted, as in FIGURE 3C, the output is a pulse. This operation turns the sample pulses of FIGURE 1A into the held pulses of FIGURE 1B. Hence, the only differences between them is the filtering produced by our delay-integrator-subtractor. The implied hold is nothing more than a filter which has the frequency response given by the equation below:

$$H(f) = \frac{\sin(\pi f_a T_s)}{\pi f_a T_s}$$

where f_a is the audio frequency and T_s is the sampling period. It is impossible to demonstrate this equation without higher mathematics and the reader should consult other references for a proof. The important thing to note is that this filtering effect reduces the high frequencies in the audio band somewhat, and this must be corrected. The filtering effect of the hold is called the *aperture*



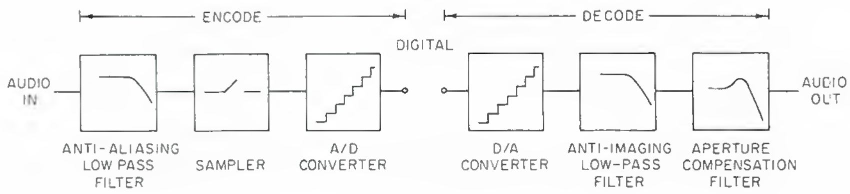


Figure 4. The complete A/D, D/A system.

response. As in photography, the aperture is the size of the opening: pulse width. With the pure samples, the aperture is extremely small and there is no effect. A compensation filter must be included in the decode process to compensate for the effect when a hold is used. The hold is implied by leaving the DAC on between samples.

GENERATION OF IMAGES

Let us return to the subject of the image frequencies and their origins. The audio signal was said to exist only between DC and half the sampling frequency. Where then did the image signals come from? In the previous article we considered the sampling process, and showed that it is able to create the same sampled signal from many different input signals. In particular, a 1 kHz sine wave and a 41 kHz sine wave, when sampled at 40 kHz, produce the identical result. This is a "mapping" of two (or more) sine waves into the same result. The digital number sequence represents both sine waves. In fact, the same sequence represents 1, 39, 41, 79, 81, 119, 121 kHz. The input anti-aliasing filter resolves the ambiguity by removing all except 1 kHz. However, the digital number sequence does not know that only one possible sine wave could have produced that sequence. At the output, the decoder produces *all* sine waves which could have created that sequence. We design the anti-image filter to allow only the one which *we* know produced the sequence.

One could have an anti-aliasing filter with a bandpass from 20 kHz to 40 kHz if one also has an anti-image filter which is the same. Such a digitizing system will only pass those frequencies, but the output frequency will still be the same as the input frequency.

COMPLETE SYSTEM

This completes the introductory discussion on the theory of digital audio systems. The complete system is shown in FIGURE 4. So far, we have done nothing with the digital word sequence other than convert it back to audio. In a real application, the digital sequence is processed in some interesting fashion, such as delay, reverberation, storage, or transmission. A digital tape recorder would record the digital sequence for later playback; a delay system stores and recalls the sequence. A reverberation system is much more complex, since it processes the sequence to create a new digital sequence.

We should also note that the encode process produces all of the theoretical degradation, by the quantization noise of conversion, and bandlimiting, by the anti-alias filter. The decode process produces no further degradation, since there is no quantization and the anti-image filter is the same as the anti-aliasing filter. The former removes audio components, whereas the latter removes images created by sampling.

In the next few articles, we will turn our attention to the actual implementation of the converters. This will introduce us to the real problems of a digital system. ■

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Voice Prints and Stress Analysis

What is a voice print? It is an everyday term for a sound spectrogram of a person's voice, continuously taken through time. Does this mean you can read a voice print? Now that depends on what you mean by "read." With some practice, you can recognize patterns that represent specific words and thus, from memory, you will know those words again on the voice print. But a great deal more practice would be needed before you could construct words from your knowledge of the way sounds go together, rather than by simple recognition.

A SOPHISTICATED "ID"

Now, voice prints have been used as a sophisticated form of identification. A person says a name and the machine recognizes if it's the owner of the name, or someone else, saying it. Only the owner's way of saying it will be accepted as "authentic" by the machine.

Now, think about listening to different people saying the same name. They all say it differently, and the voice recognition machine can detect such differences, when compared with a permanent voice print of how the person himself says it. But, given that it could tell when you say your name, or when I say mine, could it tell if it was you saying mine, or I saying yours? Only if those specific voices, saying those specific names, were programmed into it. It could not tell, at the present state of the art, merely from prior inputs consisting of each of us saying our own names.

STRESS ANALYSIS

That's a little of what voice prints can do. Now what about stress analysis? What does a stress analysis machine do? It must work with the same spectral content, but process it in a different way. It is not concerned with recognizing the words you say, but with recognizing whether you are in a relaxed or tense state when you say them.

Research has found that subtle changes in the way the vocal chords work occur with nervous tension, as compared with when the person is relaxed. What does this mean? A person's voice rises and falls in pitch, some more than others, in the normal course of speech. It may be for emphasis or some other purpose. That, in itself, does not reflect tension or relaxation. Tension tends to make a person's voice what you would subjectively describe as "edgy."

How can voice analysis tell that? Not so much by frequency content, as by frequency combination in that content, just as in music. Strident sounds result from inharmonious combinations of tones, while melodic ones result from harmonious combinations. This means that any such detector must first frequency analyze the sound, and then apply the output from such analysis to some form

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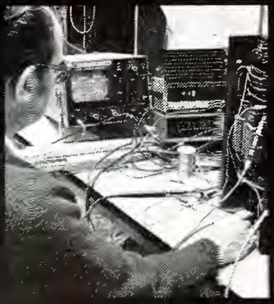

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NEPTUNE ELECTRONICS, INC.
934 N.E. 25th Avenue
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of logic circuitry, that functions on the basis, not of the frequencies present or absent, but on the relationship between the frequencies present.

If you are an experienced listener, perhaps you can tell the same thing that a voice stress analyzer can just by listening to the voice itself, in much the same way a musical conductor assesses the performance of his musicians. Advocates of stress analyzers, for the most part, want us to believe that the machine is "smarter than the average ear," with apologies to Yogi. All we can say is that a stress analyzer removes the result from the subjective realm — the opinions of various listeners — to an objective realm — a precise indication on a meter or indicator of some sort.

DON'T JUMP TO CONCLUSIONS!

But beware of jumping to the conclusion that because the result is objective, it is infallible, as regards determining whether the person being analyzed is lying or telling the truth. Tain't necessarily so!

First, any machine that will reliably detect stress, or the absence thereof, in a particular person's voice, must first be calibrated to that person's voice. Some people's voices are more strident when they are relaxed than others are when they are under tension. This is just a characteristic difference between individuals, and is no indication that one person is normally more tense than another.

So, you've calibrated the machine, using the procedure specified by the manufacturer, and you have tested it by having the subject say something both of you know to be true, and also something you both know to be a lie. Good, it indicates for this subject. Now, can you be sure the result is reliable?

The point to remember is, all it can really tell you is whether the question you ask the subject *causes* stress. And stress can definitely be brought on by reasons other than lying. In Victorian times, the word "sex," particularly used in mixed company, would have caused stress. Would the responses achieved indicate that the people present were lying about something? Not necessarily: it would just indicate that a taboo word had triggered stress. And there could be other reasons for a similar response, perhaps individual to specific subjects.

IN RELATION TO AUDIO

How does all this affect audio, and what we may be able to do in the future? It may help to get us thinking about the parameters involved. Perhaps a voice stress analyzer could be adapted to determine whether a specific person, for whom the device has already been calibrated, or someone else, is speaking in a specific piece of sound track.

On solo sounds, it would be possible to modify the same technology to determine what musical instrument is being played, and possibly, even to identify the musician playing it, from something equivalent to his "style." But that is still a long way from being able to isolate such a sound, in a complex sound channel, obtained from a full orchestra, playing at once. This is a capability a musical conductor can acquire and maybe, down the road aways, technology will find a way to duplicate that capability.

DUPLICATING HUMAN HEARING ABILITY

So far we can distinguish three basic systems, with variations, each of which partially duplicates an ability of human hearing. A voice print gives a visual print-out of any source of sound, as a spectrum of changing frequency content against time. With some skill, which must be acquired from working with them, you can "read" them. You may be able to detect a difference between the voice prints of different people saying the same thing, but you would be hard put to identify who was speaking, without such a constraint of having them say the same thing.

A second system is based on providing it with an electronic memory, whereby it can remember the precise voice print

of a particular person saying his own name, and use that memory to reject the print of anyone else saying the same name. Data on the precise voice print can be entered in a magnetic memory on a card, similar to a credit card, which is inserted in the machine, while the presenter of the card says his name into a microphone.

Thus, such a machine is a form of comparator: it compares the voice print of the freshly spoken word into the microphone, with the recorded print presented to it on the card; but it has only two results; go and no-go. If the two match, the person is identified. If they don't, he is rejected as an imposter.

The third kind of machine is the stress analyzer, often called a "lie detector." This analyzes the prints a little differently. It does not seek identity or non-identity, but rather certain qualities about the print. To do this, it must in some way be "calibrated" to the individual being "tested," usually by having him state some already verified fact, which the operator knows to be true, and then getting him to say something he knows to be false. It uses differences about the print that have been found to be indicative of the state of stress under which the person speaks.

Because it needs calibrating, it obviously has the basic capability of telling the difference between individual voices, although it is not set up to do that, like the security device is. So all three duplicate part of human hearing capability. All three start by producing a frequency analysis of sound over time, in a form that could be presented as a voice print.

THE AMAZING HUMAN HEARING FACULTY

This leads us to conclude that the human hearing faculty has an infinitely better memory system than any of them. We can hear what a person says, and if we have heard that voice before, we can also identify the person saying it. We recognize words we have heard before, perhaps many thousands of them, and can interpret them into meanings (rightly or wrongly). A person transcribing a dictated message can "translate" those sounds into visual symbols, such as those you are now reading.

Efforts are being made to produce a reliable transcribing machine. One of the main problems is that of accommodating a wide variety of different voices, still recognizing the same set of words, in a large enough vocabulary.

Human hearing accepts all signals picked up by the person's ears, and processes them. It has the ability to ignore non-relevant sounds, according to the way the person's "listening" faculty programs it, even ignoring sounds that occupy the same frequency spectrum as those being listened to, by using pattern recognition. To do this, it must have some way of telling what frequency components "belong together" as parts of the same sound: the voice of the same person speaking.

How does it filter off—or whatever would be the appropriate term—just those elements within the common spectrum that all the sounds being picked up are using, and identify them as that particular person's voice? To the best of our knowledge, this is a capability that technology has not been able to duplicate, as yet. To do any of the things we have been discussing, the machine must be provided with the sound to be analyzed, pretty well isolated from all other sounds. Does anybody have any answers to this? ■

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N. I. WEINSTOCK

db Sound With Images

Film vs. Video

• American standard television relies on 525 scanning lines of electron dots to fill a screen anywhere from two inches diagonally on up to ten feet and more, if you include the huge projection systems used in theaters. The horizontal resolution of the typical ¾-inch video system is approximately 260 lines in color and 330 lines in black-and-white. This adds up to about as many bits of information as a fast 16 mm film stock; that is, a film that can be used with available natural light. (Film's dynamic range of bright-light-to-darkness is far greater than video's, in any format compared.)

Much professional film production is done in 35 mm, creating images with a resolution far better than anything video can reproduce. Confusingly, for film production done exclusively for showing on television, producers still prefer the 35 mm format, even though this difference in quality cannot be reproduced. Why?

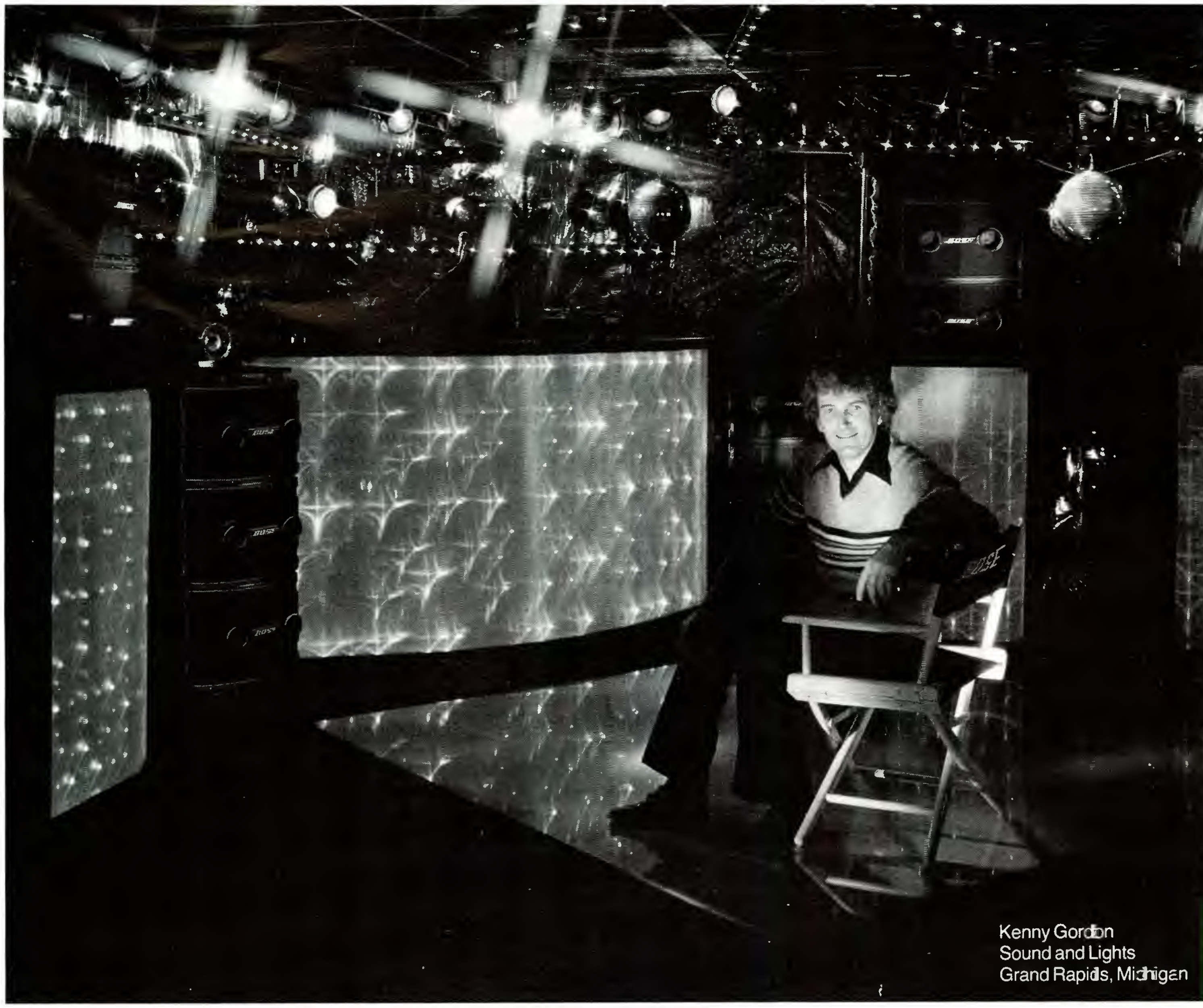
Could it be the audio portion? (Not yet a crucial part of the television experience.) Most audio on film is still rendered (perhaps "rended" is a better word) in the final product into "optical sound," the recording device invented back in the early twenties by Lee De Forest. This capacitance system has severely limited dynamic range (about 500 to 14,000 Hz), and ages quickly. As the film gets dirty or scratched, which is inevitable with hard celluloid, you hear it. Historically, film-sound has been so bad that one didn't even mind the horrors of "mag stock," or the hard, celluloid-backed, sprocketed, magnetic tape that was used to maintain sync with the pictures. The hard backing and sprocket-popping degrades the response, even before the sound gets transferred to optical. In the 16 mm format, slow transport speed also contributes its share of damage.

In contemporary film production, the final sound is often recorded on a "mag track" instead of optical, particularly in any feature film claiming to have better than average stereophonic sound, Dolby sound, etc. Still, the width of each sound stripe involved (0.063 inches) is somewhat thinner than a single track on a pro-audio multi-track tape recorder (0.070 inches). (For more on various sound track formats for film, see the March 1980 **db**—Ed.)

Of course, as I pointed out in my last column, sound on video tape is no better, and sound on television is even worse. Yet, both have high capabilities, and should improve rapidly. For the time being, however, quality of both picture and sound remains on the side of film. But the larger reason why film is used more often for programs to be televised probably lies with force of habit.

Time was, video equipment was all bulky stuff that did not lend itself to location work. Of course, time was that film equipment was zip-locked in that same bag. But film broke out of the studio about 1960, while video is just now beginning its break-out with smaller, more portable equipment. In fact, with fine video cameras about the same size as 35 mm cine cameras, and with the same degree of lighting needs, and VTRs down to manageable size, video now rivals film's convenience factor.

And, video has a few added conveniences of its own. Electronic editing has just spread to the audio world, and



"Bose® 802s are amazing speakers."

Rather than blowing our own "horn" and describing how the Bose Model 802 speaker system does away with awkward horns and bulky bass cabinets that are nightmares to use—we've decided to let one of our enthusiastic, satisfied dealers do the talking:

"For me, the basic advantages of the Bose 802 over any other type of speaker system are sound and size. Most of my customers want speakers that sound good and don't spoil the show by drawing attention to themselves.

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"Another fantastic thing about Bose is that I never worry about the 802 blowing up. That's quite a contrast to some of the compression drivers I have used. In this business, the operators often don't understand how much the systems can take and they get abusive. Bose 802s can take just about anything those guys hand out.

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most of us are familiar with the incredible advance this represents over the age of the ragged razor blade and greasy fingers. Film is necessarily still in that ragged age: the editor may use a three-screen, programmable Steenbeck editing bench, but in the end, he physically "cuts the plastic," a step unnecessary in video. In fact, here video's usually-overrated higher equipment cost is evened out: the hardware to edit video (one editing deck, controller, processing amp, and perhaps a console) is much cheaper than the finest film editing equipment. Without the finest possible editing, film's advantage in quality is thrown away. Not a problem with video.

The convenience of the ¾-inch cassette should not be underestimated. On a film set, there's usually a crew member whose sole job it is to make sure the camera's magazines are loaded; and those magazines, in 35 mm, won't usually roll for more than twenty minutes. Much more importantly, the convenience of seeing what you've got on tape as soon as you've got it is still something the medium of film can't match, though Polaroid is trying.

Film and videotape are about equally difficult to duplicate, but if the film has a magnetic sound stripe, that's an extra step—and that takes longer. Speaking of which, it is our understanding that another reason mag striping never really caught on is that films are still shipped—by mail or equally-destructive courier—in cans, and are easily demagnetized or remagnetized. Video tape would also have this problem, and therefore is a bit less convenient to distribute than optically-striped film.

But on the distribution end, there is a psychological factor working in video's favor. People are getting used to watching television, and coming to prefer it to sitting in a darkened, more formal room. At corporate conferences, sales meetings, in-house teaching, in schools, and indeed in all the domains of the industrial film, the public is coming to demand video, not film. Could this not also represent a general belief in video as more technologically advanced?

It is, of course. And if films at present record images with more quality than videotape, they record that image as well as they are ever going to. While advances such as the Steadicam, better dollies, and smaller lights benefit both video and film, the film itself, and its basic hardware, is at the top of its technology. There will never be better contrast or color represented on film than there is now, and a supportable position in debate is that the first Technicolor feature films produced better—perhaps more-interesting is the word—color than today's processes. Certainly film and film equipment will never be more cheap and available than they are today.

Video may not have the quality of the best of film just now, but it will in the future, for video is at the beginning of its growth curve. We say "video" advisably, and not "videotape," because much of that growth in quality will come from the videodisc, and eventually from direct computer storage of information.

Make no mistake about it (here comes the auspicious pronouncement), the videodisc represents as much of a superior technology to videotape as magnetic tape does to film. (The capacitance recording, stylus-in-groove, mechanical system soon to be marketed by RCA and its licensees, with all its inherent mechchanical imperfection, and with lousy mono sound, is not what we are referring to, and hopefully it will fall by the wayside soon—sorry about that, RCA.) We speak of those video systems that use the laser—probably, more than atomic energy, the greatest discovery of our time. Aside from zapping some future Starship Enterprise, the laser is the recording apparatus of Philips/Magnavox's new disc, and provides striking improvements in the recording of both picture and sound, bringing both up to the level of the highest quality film and digital audio.

Next time, we'll get back to solid engineering ground, as we unravel the mysteries of the Time Base Corrector. ■

New Products & Services

SPECTRUM ANALYZER

• The Model SA-91 Audio Spectrum Analyzer offers a cost-effective way of obtaining high resolution by offering 1/9-octave resolution at a price usually equalivent to 1/3-octave analyzers. A single, sharp voltage-controlled filter is swept over the entire audio spectrum to obtain 1/9-octave resolution in 91 bands over a 20Hz to 20kHz range. Dynamic range of display is 20dB in 101 distinct 0.2dB points. This analyzer is useful with any equalizer. The internal memory can store two complete frequency response curves.
Mfr: Spies Laboratories
Price: $1195
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GAIN REDUCTION AMPLIFIER

• The Model 3320 Gain Reduction Amplifier has only one operating control, Range Reduction, calibrated in dB. Output peaks remain constant under all Range Reduction settings once the Input and Output presets are adjusted to the desired bus level. The Model 3320 is a panel mounting unit 5¼-in. high by 1½-in. wide and 5¾-in. behind the panel. The 3320 requires a bipolar 15 volt dc supply.
Mfr: Modular Audio Products
Circle 51 on Reader Service Card

PATCH BAY

• The PB 2896 patch bay system features 16 stereo inputs and outputs with 64 gold-plate phono connectors on the rear panel and 3-conductor Bantam jacks on the front. The fully-normalled design means that no patch cords are necessary for normal system operation. Internal switches are also provided to disconnect "normals" on components not normally hooked up in the system. Able to fit into a 19 in. rack space, the patch bay offers a PBX22 stereo balanced bridging adaptor, PJ713G gold-plated patch cord, and other options.
Mfr: Audiovisual Systems
Price: $650.00
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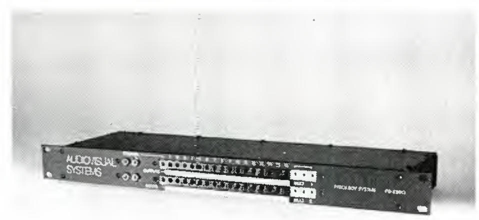
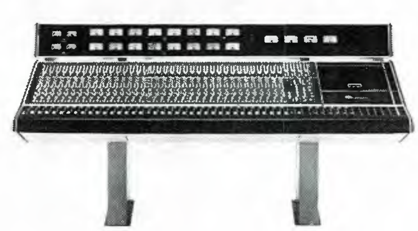


AUDIO MIXING CONSOLE

• The Series 700 marks a new line of recording consoles. The first product is a 36-in, 16-out, console with four effects outputs, two foldback outputs, stereo control room and studio monitor outputs, designated the Model 720. The mainframe contains a modular plug-in jack bay with 432 jacks. It also contains 24 VU meters that indicate all the console program outputs. Other features of the 720 include transformerless microphone preamps, three-band parametric-type equalization, and VCA input subgrouping.
Mfr: Audiotronics
Price: $41,000 to $63,000
Circle 53 on Reader Service Card



MOBILE CONSOLE

• Technics has introduced a rugged, totally mobile professional turntable console that features the quartz-phase-locked direct drive system. The Model SL-9560 is a stereo (or mono) phono disc reproducer designed for professional applications. The unit consists of two highly sophisticated, acoustically isolated sections: the turntable and tonearm/cartridge "deck" section, and the control illumination "main" section. The deck section includes a direct-drive turntable featuring a gimbal-suspended tonearm, which incorporates, a professional quality moving coil cartridge. The control section consists of a phono-equalizer amplifier with a high pass filter, tone controls and a monitor amplifier with speaker.
Mfr: Panasonic
Price: $6,565.
Circle 54 on Reader Service Card

MIXER

• The Series 200-8X2 is intended for both film and TV use as well as for all mono or stereo recording or mixing applications requiring a small, (13" x 17.5" x 7") battery-operated mixer. The 8X2 will operate approximately 20 hours from an external rechargeable Gel-Cell battery pack (supplied) or any 12 volt source. Among the inputs included are ones for phantom power, phase reverse, cue and echo sends, 4-position input gain set switch and three equalizers with 4-position mid frequency select. Two standard VU meters, ganged Duncan master fader and phones monitor are on the master panel.
Mfr: Interface Electronics
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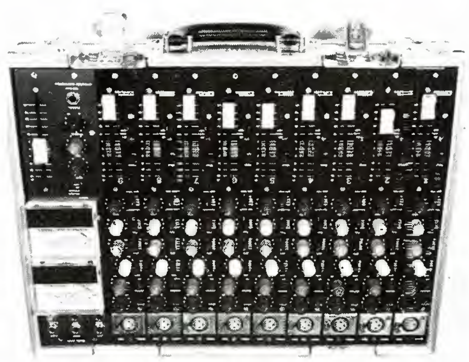
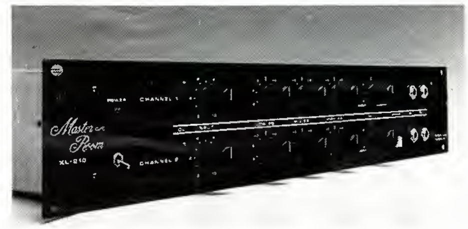


MIXING CONSOLE

• Series 4 (four output buses) is a new line of mixing consoles offering optimum features and performance for both sound reinforcement and four or eight-track recording sessions. Offered in either 12 or 20 input fully-modularized frame, some of its features include: transformerless input circuitry; three-band, continuously-variable equalization in each channel; peak LED and 20 dB pad on each input; eight independent returns, PFL, and six bus assigns through submasters (4) and R/L stereo buses; and full provision for multi-track monitoring and assign.
Mfr: Tangent Systems, Inc.
Price: $2800.00
Circle 56 on Reader Service Card

REVERBERATION SYSTEM



• The Master-Room XL-210 incorporates a recent technological breakthrough that provides the highest quality reverberation at an affordable price. The XL-210 is a self contained 3½-inch rack mount unit that features two completely independent stereo channels that are easily switchable to monaural operation. Input and output connections are via ¼-inch phone jacks located on both the front and rear panels. This unique feature allows convenient break-in patching at the front panel without disturbing the permanent rear panel connections.

Mfr: MicMix Audio Products Inc.
Price: $950.
Circle 57 on Reader Service Card

SIGNAL PROCESSING

• A new modular signal processing system, the 900 series which includes a noise gate, de-esser and compressor, was recently introduced. The series features convenience with flexibility. Up to eight signal processing modules can be fitted into a rackmount unit measuring just 5¼-inches high. Designed for fast installation, standard connectors enable the rack to be wired easily into a system. The interchangeable signal processing modules slip in and out in seconds.

Mfr: dbx, Inc.
Circle 58 on Reader Service Card

db Editorial

NEXT MONTH, the National Radio Broadcasters Association holds its annual convention (Los Angeles, Bonaventure Hotel, October 5-8). In honor of the event, we devote this month's **db** to a look at various aspects of broadcast audio.

Usually, we tend to think of pro' audio as something that happens in the recording studio. After all, recording engineers are into audio, while broadcast engineers concern themselves with, well, broadcasting.

Broadcasting what? Why, audio of course; for example, your very next chart-buster: the one that hasn't got a chance unless it gets plenty of air play.

After recording, mixing and mastering, your potential hit record will surely encounter some "we'll fix it *after* the mix," when (and if) it arrives at the broadcast studio. Some stations have their own special "sound," while others simply try to be the loudest game in town. In either case, your priceless master is in for even more processing, which will hopefully keep those listeners tuned in.

But how much processing? Eric Small has come up with a means of "modulation analysis" in which he quantifies what goes on just before the transmitter. Using the New York marketplace as his guinea pig, he makes some interesting comparisons between stations from his lab, somewhere in the wilds of darkest Brooklyn. Barring an offer he "can't afford to refuse" from some program director, Eric will probably show up in your home town, as he goes about Measuring the Sound of Radio.

Playing a significant role in the sound of radio is the celebrated (?) NAB cartridge. Recording engineers, long-accustomed to the relative sophistication of open-reel, may wonder about the continuing popularity of the cartridge system in broadcasting. Elsewhere, the cart has all but disappeared, in favor of cassettes. Why not in broadcasting?

Why not indeed, asks Eumig USA, Inc. Technical representative Tom Bensen, who describes the company's micro-processor-based system, which may be used to control up to 16 cassette decks, using a low-cost personal computer. With a little creative imagination, such computer-controlled cassette systems will probably start showing up outside the broadcast studio as well.

Speaking of outside the broadcast studio (nice segue, huh?), what about wireless transmission inside the recording studio? The wireless microphone has been around for some years now, but until recently it has been regarded as somewhat of a compromise, to be used only when microphone cables are absolutely out-of-the-question. However, wireless transmission and reception has at last come of age, and John Nady offers us an interesting overview of theory and application, both in and out of the recording studio.

While wireless microphones are finding their way into the recording studio, noise reduction systems are showing up outside the traditional studio venues. Dolby film sound comes to mind, and of course there's the Dolby 'B' system in consumer audio. Now, dbx is beginning to make its presence felt in broadcasting, and has in fact installed a custom-designed noise reduction system in National Public Radio's satellite communications system. This month, dbx chief engineer Leslie Tyler, shares with us the dbx approach to broadcast signal processing, which naturally includes a look at noise reduction, plus some talk about actually expanding dynamic range.

We close with an inquiry into What's New at the FCC. As you may have heard, there's a recession going on out there, and its having its effects on all of us. Traditionally, the recording/broadcast industries have been comparatively "recession-proof," as people stay home more and therefore spend more time listening to radio and records.

Well, this time around, our industry is also feeling the pinch, and it could certainly use a little perking up. In short, this would be a good time to see AM stereo and FM quad finally come off the drawing boards. The recording industry would be revitalized, AM radio would come a lot closer to "hi-fi," and FM would be able to surpass anything heard to date.

However, first we must contend with the Federal Communications Commission. The Commissioners rarely act in haste. Some say they rarely act, period. Our Special Report will bring you up-to-date on the Commission's most recent contributions to the state-of-the-art.

In a New York Times (July 20) article entitled "In the Public Interest. Hah!," Jerome Gillman (WUST-FM) describes his own seven-year battle with the Commission. He concludes with the suggestion that the Commission must be re-structured from scratch. The NRBA has sent reprints to all members of Congress, and it asks interested parties to write directly to their legislators to convey personal opinions and experiences in coping with the FCC.

Depending on the results of the November election, there may be some changes at the Commission, as well as elsewhere of course. (Some Washingtonians are already beginning to pack.) And so, this might be as good a time as any to let your elected representatives know how you feel. Maybe something good will come of it—but not unless you do your bit. ■

TOM BENSEN

Computer-Interfaceable Eumig FL-1000

With all the problems confronted in using an NAB cartridge system, why not give cassettes a try?

In broadcast audio, the NAB cartridge system is by now a well-established fixture. Therefore, before going into the technical details of interfacing a cassette system with a micro-computer for broadcast applications, it might be wise to answer the question; Why use cassettes in the first place?

Quite simply; Why not? The broadcaster has been living with the NAB cartridge for more than 25 years, trying to cope with problems such as excessive wow and flutter, poor stereo separation, phase shift and imaging. The graphite-lubricated tape requires constant maintenance and its residue causes jamming, tape breakage, etc. Because of all of the above, the NAB cartridge also suffers from instability of frequency response.

Though advances have been made recently in the NAB cartridge system, it is still unable to compete with the specifications of today's cassettes. According to Ed Havens, technical services manager at TDK Electronics, "It is widely recognized that more emphasis has been placed upon the audio cassette in the last ten years than any other analog tape medium. Improvements in all areas of their development have elevated cassettes technically, certainly exceeding those of the NAB cartridge, while effectively challenging open-reel recorder specifications at 7½ ips. The most significant improvements in cassettes are generally noticed in two areas: tape formulations and shell construction.

New tape formulations such as Ferric Cobalt—an equivalent to CrO_2—and most recently, pure metal particle tapes, have measureably improved the high-frequency headroom, and extended overall response and signal-to-noise ratios to meet or exceed broadcast specifications.

Improvements in quality shell construction have helped reduce wow and flutter and maintain near-perfect alignment between the A and B sides of the cassette, while smoothing out tape travel within transport mechanisms in general. In short, cassettes have come a long way. They work better, they certainly sound better, and have proven to be a reliable medium."

With this in mind, it was Eumig's intention to design and build a microprocessor-controlled tape transport mechanism which would fully exploit the cassette in terms of its technical superiority, while providing the broadcaster with an entirely new concept of automated program operation. As the design of Eumig's FL-1000 cassette progressed, three basic goals were established:

a) provide a means to reach any position on tape, quickly and precisely.
b) engineer a versatile and convenient remote control system.
c) create a data bus system which would enable an external intelligent device (computer) to communicate simultaneously and bi-directionally with several sub-systems.

To summarize, the FL-1000 design allows it to be addressed via computer, be given a command, execute that command, and, move to the next event.

In fact, up to sixteen FL-1000s may be connected at once, suggesting obvious applications to the broadcaster. In an automated application, several decks (perhaps four to six), can be devoted to music playback. Two or three decks may effectively air and maintain the spot clusters, and time announcements can be accomplished by another deck. One would program all announcer breaks, another would be used for all P.S.A.s, while two decks would be used in the production area. One deck might be used strictly for automated network joining, for delayed newscasts, sport feeds, etc. In addition, one deck might be devoted to the local newsroom. All decks can be random accessed, playing any selection in any order.

At the control center is a relatively inexpensive computer, with at least 32K of memory. For sequence storage and programming, a floppy diskette drive may be used, and finally, a printer will notate automated logging of the broadcast material.

In the broadcast environment, such a system will provide full random access to any point on tape. Cue tones are not necessary, as the FL-1000 uses an optically-sensed counter which is universal among all FL-1000 decks. Due to the transport design, which will be discussed later, this counter system is accurate to the digit, regardless of the number of plays. It is therefore possible to play selection 2, advance to selection 7, rewind and play selection 3, etc. This is accomplished via a several-step process, which begins by using the FL-1000 as a digital recorder.

By means of three command codes (data onto tape, data from tape and data onto tape and monitor); digital tape may be recorded onto the tape via the bus lines and re-read.

Tom Bensen is technical representative for Eumig USA, Inc., Great Neck, NY.

All transmissions fed through the bus using a ten-pin plug/cable arrangement, which contains the following:

Line Designation	Pin		
Code Line	8	RFT	Ready for transmission
Handshake Line	7	BY	Busy line
Handshake Line	6	H_1	
Handshake Line	5	H_2	
Data Line	1	D_0	negative logic
Data Line	2	D_1	negative logic
Data Line	3	D_2	negative logic
Data Line	4	D_3	negative logic
Ground	9		*see note
Power Supply	10		+5 volts (TTL)

*To avoid transmission disturbances by inactive machines.

For the purpose of a broadcast demonstration made by Eumig at the last NAB convention, material was assembled in the following manner:

During recording of broadcast material, counter locations were noted and typed onto the computer CRT via the terminal keyboard, thus creating an Index or Table of Contents for that particular cassette. When fully assembled, the FL-1000 is told to write it (digitally) onto tape, by entering the command code, 1001.

The FL-1000 recognizes and confirms the command, switches into recording mode, mutes Audio in and out, and signals the receipt of data by illuminating the Source, Tape and Record LEDs. Data line D_2 becomes a serial data input (TTL signal). At this point, the CPU can transmit data onto the cassette via this (D_2) line. During data recording, BY, H_1 and H_2 remain in force, thereby blocking access by other participants on the bus. The data recording is terminated when the CPU frees the bus line. The FL-1000, in turn, frees the bus and stops.

The functions "data from tape" and "data onto tape & monitor" are treated identically; the data read from the tape are transmitted via D_0. The functions "data onto tape & monitoring" serves to recognize the end of the leader at the beginning or end of the cassette. The user can select the coding. A self-clocking code with approximately 2,400 Baud is recommended. The "data from tape" function may also be used to search intervals between pieces of music. During these intervals, the FL-1000 can also be used to load programs into the computer.

The information brought through the bus—in this case the Index—is recorded onto tape. When requested, the digital data is transferred back into the memory of the computer. All that need be done as a function of the program is to request a machine number and a selection number; for example, machine 1, cut 2. The FL-1000 will then advance towards or rewind to that location and execute the command. This is possible through the rather unique transport mechanism. Fabricated from die-cast magnesium alloy, it is a totally unconventional design which uses no belts, pulleys, flywheels or solenoids. It is a single-capstan, two-motor drive system, which uses a light-weight (under 7 grams) opto-electronic capstan and disk arrangement to both reference its speed (15,000 times per second) and maintain reduced wow and flutter—quoted at 0.035 percent WRMS or less. There is virtually no mass to overcome in attaining proper operating speed of 1⅞ ips, which will occur from dead stop within 40 milliseconds.

As indicated, the FL-1000 can seek a counter position in response to a manual command from the user, or via the computer interface. The counter system, which operates from the take-up spindle, is also driven by a separate optical reference within the transport, when given a selection number by the computer from the index in memory. For that cassette, the FL-1000 "knows" to go to the appropriate counter position.

In the foreground, a Commodore PET micro-computer handles programming chores, keeping track of ten or more cassette decks.

The most probable concern, and most often asked question is, what if the counter location changes due to tape stretching or aberration in tape packing density? There would be valid concern if addressed to a conventional dual-capstan or single-capstan design which utilize flywheels. However, the FL-1000 transport does not change packing density through normal play, fast-forward, or rewinding of the cassette, due to the reduced inertia placed on the cassette mechanism. Therefore, not only do the tape counter positions remain accurate, they are universal among all other FL-1000's. For example, if a selection begins at location 2248 in machine 3, and that tape is removed, placed in another FL-1000, rewound to the beginning, and told to go to 2248, it will still be in virtually the identical location (±0.30 seconds). This tolerance is certainly more acceptable than the lead time needed to "punch-up" a cartridge or that which is necessary for an open-reel to attain proper speed and have the tension arms settle in place.

At this time, there are quite a number of stations in the final stages of incorporating the FL-1000 into on-air operation. At the vanguard is Associated Communications Corporation, an 11-station chain, based in both Pittsburgh, Pa and Tampa, Fl. According to director of engineering, Will D'Angelo, "Our goal was to incorporate the FL-1000 in a series of steps within our system. The first step was to incorporate it into the system as a direct replacement for the cartridge machine. Secondly, we were able to effectively interface it into a Cetec Schafer (System 7000 Automation System) as a music playback system. The third step was to use a computer for simple sequencing and operation. Finally, we are nearing completion of a full random-access system, using the FL-1000 for all phases of operation, including interfacing it into our station business system."

D'Angelo explains that they have been able to accomplish the business system interface by using a frequency-shift keying (FSK) track on the right channel and commercial material recorded on the left channel in monaural for decks devoted to spots. Therefore, commercial logging is accomplished through reading the FSK track and in turn, is interfaced with the station business system to generate direct billing.

In addition to Associated Communications, there are other broadcast facilities throughout the U.S. in the process of incorporating the FL-1000. Some will be as elaborate as Associated's full random access system, while other planned applications are for simpler, live-assist environments.

Subsequently, there are other applications for the FL-1000 outside the realms of broadcasting, with new ideas being suggested daily. Some possible examples are:

—Organization and rapid access to large archives of sound effects and sound recordings, etc.

—Medical applications for the storage and quick retrieval of data and procedures in the operating theater during surgery.

—Automated time-indexed public address systems in industry, hospitals, and institutions.

—Text storage in libraries with quick access to selected passages, recordings, and other stored information.

—Multi-media presentations, multi-vision, etc., with simultaneous control of light and sound effects.

—Automated audio editing using several slave decks as sources and assembling excerpts from each to create a master (a similar concept to Umatic video editing in ENG operations).

—Educational and instructional audio-visual training.

There are undoubtedly a multitude of other applications which are yet to be suggested. It appears too, that the versatility of applications and functions of the FL-1000, when interfaced with a computer, are solely a factor of the limitations of the softwave involved. ■

JOHN NADY

Wireless Microphones— On Stage and in the Studio

On stage and in the studio, the performance of wireless microphone systems may now equal or sometimes surpass conventional methods.

WIRELESS TRANSMISSION OF SOUND was first achieved around the turn of the century. However, only within the last several years has the evolving state-of-the-art finally yielded radio-transmitted audio with both the fidelity and dynamic range required for applications subject to the most critical ear. This achievement had long been an evasive, yet natural, goal for radio transmission.

Interestingly (and fortunately), the impetus for the development of this high-level audio performance also dictated the need for simultaneous development of a highly-miniaturized radio transmission system. As it turns out, the screaming rock-and-roll lead guitar places more demanding dynamic range requirements on a transmission system than any other audio application. As a result, one of the most significant breakthroughs in radio-transmission audio performance was spawned in the course of developing a miniaturized wireless microphone-type system for playing electric guitars without cords.

Although wireless microphones have been commercially available for over 20 years, hard-bitten longtime users can testify that—until recently—performance was never even closely comparable to that of hardwired cords (especially for use in demanding applications such as cordless electric guitars).

The live entertainment and recording sectors of the general audio market can most immediately reap the significant benefits afforded by the imaginative application of this newly available technology. Of the two, the live entertainment field has a sizeable head start in advantageously utilizing wireless. This is because of its past willingness to compromise its audio performance, as evidenced by the relatively widespread use of earlier-generation wireless microphones. It is inevitable, however, that the recording industry will become more aware of the new possibilities now available in wireless systems which make no discernible sacrifice of audio quality. Already the vanguard innovators are beginning to pave the way in exploiting some of the advantages of going cordless in the

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studio. Since these advances and possibilities are still new to most interested parties, I will present in this article an overview of the entire present wireless phenomenon as it applies to both the entertainment and the recording industries. I hope this overall picture will create a better understanding of the exciting promise that wireless microphone systems offer for stage and studio.

USING WIRELESS SYSTEMS ON STAGE

The advantages of going cordless in live performances basically fall into two categories: practical/aesthetic, and technical. Under the former, there are: enhanced on-stage mobility, greatly increased theatrical possibilities, a streamlined, uncluttered stage appearance, and the performer's ability to hear the total sound offstage—during both sound checks and performance. Of course, the most practical benefit of cordless operation of electric instruments and microphones is the total elimination of potential electric shock hazards. Such common shocks, caused by improperly grounded amplifiers and/or other equipment malfunctions, have even proved fatal at times (Les Harvey of Stone the Crows and Keith Relf of the Yardbirds were both electrocuted in this manner.)

A well-designed wireless system also offers several technical advantages over a hardwired cord. Cables are not ideal transmission systems for audio signals. Due to their inherent capacitance, they act as lowpass filters and can severely attenuate the highs when used with high-impedance musical instrument pickups. Although cables vary somewhat in this characteristic, most will start to "muddy" the sound at lengths over 20 feet. In addition, cables act like antennae and pick up spurious noises from light dimmers, radio signals, power lines and other external sources. Used with certain combinations of effects pedals or amps, cords often pick up nearby radio stations and CB'ers. All these things considered, it is no surprise that the list of the hundreds of now cordless top acts reads like a "who's who" that encompasses a broad spectrum of popular music, including such diverse styles as that of the Rolling Stones and Herbie Hancock. However, the natural education and emulation process has, especially in the last year or so, filtered through to the mass market, and a healthy rapid growth is presently evident in this sector of the wireless field. The use of these new wireless systems is not limited to microphones and electric guitars. Musicians of all kinds are discovering the joys of going cordless. Increasingly seen and heard on stages everywhere are cordless electric saxes, trumpets, flutes (with pickups or tiny microphones), electric violins, mandolins, acoustic guitars (with pickups) and even an occasional exotic application such as miniature drum synthesizers. It is clear that possibly within several years—within a decade for sure, as the understanding of this technology continues to spread—cordless operation of most on-stage musical instruments will become the norm.

USING WIRELESS SYSTEMS IN STUDIO RECORDING

Wireless systems are increasingly finding their way into the recording studio, and this is not surprising, considering the diverse ways in which cordless operation can aid today's recording engineer. In fact, in the last two years, there have probably been a dozen popular recordings released (gold or better) which have utilized some form of wireless operation during their production.

Before citing studio uses of wireless, I would like to briefly discuss our own experiences in this area. Most of these applications are ones I have used over the last five years or so in my 8-track basement studio. Since we had been devloping a new generation of wireless systems during this period, I had prototype new-generation wireless systems available for several years before we introduced them on the market in 1977. In fact, we have done completely cordless 24- to 32-track recordings (via extensive ping-ponging) utilizing all the wireless techniques outlined below.

Of course, our recording work was necessarily limited in time and scope compared to larger commercial studios. Recently, we have started work on an in-house 24-track recording and videotaping studio, which will be equipped so that we can expand our experiments in recording entirely wireless. In addition, although we haven't done an extensive survey of the present state-of-the-art, we have received quite a bit of feedback about studio wireless use in the last two years from customers, interested studio people, and the like, which will aid in these experiments. My own personal feeling is that there are still advantageous applications of wireless that haven't been discovered—and we hope to find and utilize them.

Originally, we were committed to exploring the concept of the totally cordless studio because of our interest in wireless systems. Not every studio owner will see the need to immediately commit to cordless studio operation to the extent we have. However, from what we have already learned, wireless use in the studio, as on the stage, should not be thought of simply as a "gimmick" to attract customers (although it has obvious potential in this regard). Rather, studios interested in upgrading their overall operations (a necessity in this increasingly competitive field) should seriously study the ways of integrating this potentially valuable tool for what promises to be its maximum benefit—a powerful aid toward improving operational efficiency.

The immediately obvious studio application of a wireless system is to allow the electric musician the freedom of physical separation from his amplifier. In the studio, this introduces a whole new dimension of possibilities. For example, physical separation from a loud amp allows a musician to hear his headset mix at lower, more manageable, less fatiguing SPLs—a definite advantage on long sessions. The musician can even play from the control room, which can often give him a better "feel" than loud headsets in the studio. Also, the logistics of over-

Rod Stewart in concert using a Nasty Cordless microphone.

dubbing on many takes can often be made easier with the engineer and musician sitting side-by-side in the control room.

In my own experience, I have even done dubs from out-of-doors in my back yard (on exceptionally nice days)—while beaming the signal to my amp and communicating with my engineer via two-way wireless systems. Exploring such possibilities can be a refreshing change from the everyday rut of studio procedure and the wireless system makes it very easy. This can be important these days, as more and more studios discover that artists are considering ambience and mood of the places where they choose to record.

There are several technical advantages of using a well-designed wireless system in the studio. As discussed earlier, there is the freedom from high frequency cable loss, and from spurious audio degradation. Another happy advantage of using a wireless system in a studio, especially for electric guitars, is an up to 20 dB improvement in immunity against 60 Hz hum pickup. In other words, a well-designed wireless system is actually quieter than a cord! Furthermore, wireless receivers have two outputs—with separate line and mic-level signals—thus eliminating the need for direct-box interfacing by allowing both amp and direct feeds simultaneously.

Of course, the ultimate advantage of cordless studio operation is probably "streamlining." The recording studio is even more a maze of jumbled cords than the concert stage, and a significant percentage of valuable studio time is lost in dealing with this labyrinth during all phases of a typical session. As in our previous 8-track studio, all inputs in our new 24-track studio will be on separate radio frequency channels. This greatly facilitates setup and breakdown. During a typical setup, we simply choose the mic, attach it to the appropriate transmitter and place it in the desired location. This procedure is an especially welcome labor shortcut on complicated multi-channel takes. And, it's a pleasant change to walk around during a recording session without having to watch out for the tangle of cords underfoot!

It is also advantageous to eliminate still more cords in the studio by using a wireless headset monitoring system. We use our tunable FM transmitters in conjunction with high-quality FM headphones. These work well for most applications without any significant sacrifice of audio quality.

Sometimes, special recording needs arise where the use of wireless monitoring is practically mandated; for example, by percussionists when there is to be a lot of quick movement by several musicians between percussion instruments. The movement is usually made extremely difficult by the entangling of headset cords. For a specialized application, such as a movie sound track, separate input circuitry on the transmitter provides for proper click-track cueing.

There is another advantageous studio use of the professional, well-designed FM tunable transmitter. It can be used in conjunction with several miniature FM portable radios simultaneously as an aid to gauging the sound on typical low-priced consumer equipment. This function is presently fulfilled rather inaccurately by the use of small "car speaker" type monitors, generally driven by atypically overpowered, overly-clean amplifiers.

USING WIRELESS SYSTEMS IN LIVE RECORDING

Live recording can also benefit from the judicious use of wireless systems. In such situations, the extra cable runs needed can be a logistics nightmare. Not only are there more microphones on stage adding to the tangle, but long, difficult cable runs to the recording track are often needed. Sometimes, special staging needs entirely preclude the use of cords. We worked on one project that is a perfect example of this—the

The Nady Pro 400 and Pro 500 and Nady Cordless.

acoustic side of Neil Young's recent sound-track album from his movie, "Rust Never Sleeps." The simultaneous use of four wireless channels (two on voice and two on acoustic guitar) allowed Mr. Young to wander about the stage unencumbered by cords. The enthusiasm of the engineers working on this project surely means many more such productions will be forthcoming. Such a quality recording would not have been possible just a few years ago with the previous generation of wireless microphones.

Another logistics aid of wireless during live recording is in use with remote microphones over the audience for applause pickup. Often, hardwiring a single such microphone over high beams, etc., can take hours. From our experience, a less expensive wireless mic with less dynamic range than the top available equipment works quite well for this application.

VIDEO OR FILM SOUND RECORDING

Video or film sound recording is the area of live recording where the new generation of wireless microphones will make their greatest impact. Although wireless mics have been used for years in this classic application, their generally poor sound, spurious RF noises, and unreliability made them a mixed blessing at best. The old-style wireless systems had limited signal-to-noise ratio, which necessitated the use of compressor/limiters. Movie sound men suffered with all these audio degradations for years. In recent years, more and more attention has been placed on high-quality, clean, noise-free sound tracks (witness, for example, the increasing use of Dolby noise reduction). It is especially fortunate that the best of the new breed of wireless microphones can guarantee studio-quality sound with noise-free, wide-dynamic-range transmission, even in the most difficult shooting locations—and with no annoying compression/limiting.

GUIDELINES FOR JUDGING WIRELESS PERFORMANCE

A question remains—Does the wireless microphone impose any compromises? Certainly, the specs of today's state-of-the-art wireless systems meet the objective criteria of present recording requirements. However—as with all electronic signal processing equipment—subtle colorations are introduced that some may swear by, and others, at. Ideally, each interested user will experiment and answer this important question for himself.

As discussed earlier, not all wireless systems perform similarly. In fact, there is now a wide diversity in performance on the wireless market. How then does the potential user choose a system in the confusing and mysterious arena of cordless radio transmitters? By what guidelines does one judge a wireless system?

In order to understand the criteria by which to make an intelligent choice, be aware of the potential problems exhibited by wireless systems. These include: radio signal dropout (null spots); interference; transmission range limitations; audio degradations, including limited signal-to-noise (dynamic range); and poor reliability. A well-designed wireless system is one that has maximized performance specs with respect to each of these potential pitfalls. Let's discuss these problems.

DROPOUTS (NULL SPOTS/DEAD ZONES)

While passing through the air from transmitter to receiver, radio waves generally encounter a maze of reflective and absorptive obstructions. When radio waves bounce off nearby surfaces and meet in space such that one wave's crest encounters a reflected wave's trough, a 180-degree cancellation occurs. A receiver antenna located at that point in space will register no received signal and a radio dropout will result. Although these null spots can occur at any receiver distance from the transmitter, they are fortunately very infrequent up to ½ to ⅔ of the system's ultimate range at any given location. Nevertheless, in most critical applications, a maximum null-spot-free range is required. There are two methods of solving this problem currently offered by wireless manufacturers—antenna diversity and receiver diversity. These two methods are in no way comparable in effectiveness.

Antenna diversity is actually a misnomer—it is no more than antenna combination. In theory and practice, antenna diversity has proven to be ineffective in significantly eradicating null spots and improving range.

The only reason for the existence of antenna diversity units on the market is that they are a lot simpler and cheaper to produce than receiver diversity systems.

In receiver diversity, two separate receivers and two separate antennae are employed to process the single transmitted radio signal. In a well-designed true-diversity system, a totally silent comparator circuit continuously monitors the received RF signal strength of both receivers and instantaneously selects the audio output of the receiver with the stronger signal. In this way, there can be no vector cancellation of the received signals as in antenna diversity, since the two antennae are completely separate.

The US Navy researched the problems of null-spot dropout in its critical radio applications for years and concluded that post-detection diversity receiver switching was the only possible way to mathematically eliminate dead zones. Although receiver diversity systems are more costly than a single receiver with several antennae, the dramatic improvement in reception reliability is well worth it.

INTERFERENCE

Unfortunately, there are not yet any radio frequencies set aside by the Federal Communications Commission in the United States (or by similar radio governing boards elsewhere) solely for wireless systems. Consequently, a potential user should be aware of the frequency allocations in the locales in which he plans to use his system(s) to avert any potential interference problems.

There are two frequency bands most successfully utilized by today's professional wireless systems: the commercial FM band (88-108 MHz), and the VHF business and TV channels band (150-216 MHz). The wireless systems operating in the commercial FM band are tunable so that they can be tuned to blank spots between FM stations. The VHF systems are all fixed-frequency, and cannot be tuned to open frequencies. For applications such as traveling musicians, live recordings, etc., where clear channel accessibility and freedom from random interference is a must in all locales, a well-designed, frequency-stable tunable system, is recommended. Even for fixed location use, such systems offer a maximum of immunity from any possibility of interference from CB'ers, cabs and the like since such business bands are so far removed from the commercial FM band.

The Rolling Stones in concert. Note the Vasty Cordless on Richards' back.

Designing a usable VHF system is complicated by the fact that, since it is not tunable to an open channel, it has to be as immune as possible from weaker signals on the same frequency and from strong adjacent business or TV channel interference (i.e., a good capture ratio of 1 dB or less and at least 100 dB image and spurious rejection). Therefore, to compete with a tunable system's usability in all locales, the VHF system must be extremely well designed. With careful selection of frequencies, VHF systems do offer the advantage of enabling the maximum number of channels to be operated in one location. For this reason, in the 24-track studio we are currently building, we plan on using more VHF systems than FM tunables. When finished, this studio will probably have more radio mics operating in one location than any previous setup ever attempted! The importance of proper design to prevent any radio crosstalk between adjacent operating units is especially critical in any such multi-channel application.

TRANSMISSION RANGE

A well-designed wireless system generally has a usable range of about 150 feet minimum under adverse conditions, and up to 1,500 feet line-of-sight. Receiver diversity (two receivers with separate antennae) provides greater usable range than single receiver or antenna diversity operation, all other conditions being equal. Different applications, of course, require different operational ranges. For on-stage use, a wireless system must provide a totally solid radio link between performer and amp for at least 100 feet. Due to sound travel, there is an acoustic delay of about 100 milliseconds at that distance. It is difficult, if not impossible, for musicians to keep in time at longer distances and longer acoustic delays. For studio use, these delays are generally not a problem.

For all the applications we've discussed, wireless systems which provide at least 150 feet under the most adverse conditions will serve quite well. Not all units presently available

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can reliably deliver this level of range performance. Published specs can often be misleading in this respect and it is highly recommended that one test a system for the range necessary in any expected application.

AUDIO LIMITATIONS

The greatest variation in performance features between wireless systems available today is found in transmitted audio quality. As a result, the wireless consumer can choose between available products (often priced the same) whose audio performance varies as much as night and day.

A word of caution: since most wireless systems are a relatively expensive investment, one should do careful market research before choosing. Listen carefully to the several top choices available. Even among those companies now offering new, improved systems, there is still significant performance variation evident. Some manufacturers are obviously further along than others in developing and implementing the new technologies.

Distortion, frequency response, and dynamic range are the three key parameters in measuring the audio quality of any electronic processing equipment. Let's consider each of these criteria with respect to the operation of today's wireless systems.

DISTORTION

Measuring distortion in any piece of electronic gear can be a numbers game and wireless systems are no exception. Published specs often quote a figure that represents a best-case situation that may not reflect realistic in-field usage. Under typical conditions, most wireless systems will yield an approximate THD of 0.7-to-3 percent, depending on the transmitted audio level and the audio frequency. The best new-generation systems improve that performance to about 0.2-to-0.6 percent THD. Discerning the desirability of the subtle colorations introduced into the transmitted audio at these lower distortion levels becomes a subjective decision and depends on the application intended.

FREQUENCY RESPONSE

Frequency response in today's professional wireless systems range from 100-7,000 Hz (±3 dB) to 20-20,000 Hz (±3 dB). Again, the intended usage can dictate the frequency response needed. Published specs tend generally to be overly optimistic, so for critical applications, a careful listen or on-bench testing is recommended.

DYNAMIC RANGE

The greatest single breakthrough in the performance of wireless systems is the recent dramatic improvement in dynamic range offered by the best of the new-generation units. Signal-to-noise has improved from a previous high of about 65 dB, to over 100 dB. By comparison, a commercial FM station only registers about 70 dB signal-to-noise through a high-quality receiver, and NASA's ultra-sophisticated deep-space network only about 90 dB signal-to-noise.

It was discovered early in the development of wireless systems for playing cordless electric guitar that about 100 dB signal-to-noise was needed for immunity from annoying background hiss when played at full volume. Earlier systems (all of which are still on the market) relied on compressor limiters on the transmitter input to handle the wider dynamic range excursions regularly encountered in typical usage. This results in the unnatural "squeezed" sound on the louder signals so typical of limiter operation and makes such systems barely adequate for most applications. The best new extended-dynamic-range systems have no need for compressor limiters, and are thus comparable to hard-wiring in accurately handling any of the loudest audio inputs.

Given the highly competitive atmosphere in today's wireless arena, much secrecy still necessarily surrounds the exact details of the technology which effected this vastly increased signal-to-noise. It is well-known, however, that the key process used is companding. Of course, companding has helped to quiet audio processing systems in the past, but its successful application to wireless systems awaited the overcoming of many formidable obstacles, including unacceptable audio degradation and the restraints of miniaturization.

Schematically, the process is similar to the familiar use of noise-reduction companders in recording, where encoding and decoding enables audio reproduction through a noisy medium (recording tape) without retention of that medium's inherent noise. In this case, encoding (compression) at the transmitter input is matched by complimentary decoding (expansion) at the receiver output. The noise of the radio transmission process, which previously limited the final output signal-to-noise, is entirely circumvented. Since lower level radio "buzzes" and "whooses" typical of weakly-received RF signals are also eliminated in this manner, both effective usable range and increased immunity from null spots is also significantly improved.

Of course, anyone familiar with the different companding schemes in the recording process will be aware that audio problems can be introduced by the use of improperly designed companders. And so it is with the present state-of-the-art in wireless systems. Significant performance variation is evident among products offered by the several competing manufacturers that have so far introduced wireless systems featuring extended dynamic range using companding. For example, the signal-to-noise ratio in these differing units varies from about 80 dB to slightly over 100 dB. Also, each design exhibits varying success in dealing with the audio degradations most common to the compander: "breathing" (annoying auditory changes in the background hiss level), "pumping" (inaccurate input-output signal tracking), and transient response, especially to low-level signals. Since some companies have been working at the development of this technology a lot longer than others, this situation is not surprising.

Again, especially for the most critical intended applications, a comparative listen to the different systems available under worst-case conditions will prove invaluable in making an intelligent choice.

RELIABILITY

Like any piece of very complicated electronic gear, wireless systems are subject to breakdown from parts failures, rough handling, or just poor production and design. Each manufacturer has his own approach to solving this potential problem. The ultimate test of reliability is proven ruggedness and roadworthiness. Because the market is still so relatively undeveloped for many of the applications discussed above, the potential customer presently has only one way to gauge this very important factor in deciding which system to buy. Only people who have been using the various wireless systems for a reasonable length of time can testify to a brand's typical reliability. The primary users of the new generation of wireless systems so far have been the top touring pros. The potential consumer would do well to find out what wireless systems these artists are consistently using—he can generally be sure that their professional sound crews have researched the market carefully.

* * *

As a longtime advocate of the use of wireless systems, it is especially gratifying for me to witness the ever-broadening interest in this technology. With the dramatic improvements in performance now available, I feel the widespread acceptance of wireless systems is just around the corner. The timing is now just a matter of education. A continuing healthy growth in the general awareness of the diverse possibilities of the new wireless technology will hasten the day when wireless systems will take their rightful place as a valuable tool in all phases of the entertainment and audio industries. ■

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LESLIE B. TYLER

Broadcast Signal Processing: The dbx Approach

Presenting a brief overview of methods to keep it louder, quieter, cleaner, compressed-expanded. Why, there's nothing to it!

BROADCASTERS TODAY face a persistent problem; to maintain a consistently high level of audio quality when switching from live announce mikes, to records, to remote coverage. The "sound" of the live mike in the studio is excellent, but the phonograph records are marred by scratches and surface noise; the cartridge's high end is strangled in dropouts and high frequency saturation. Remote feeds are replete with hum, hash, and other garbage picked up over land lines.

One simple solution is to reduce the quality of all the programming to the lowest common denominator; simply run the studio audio channel across town on a telephone line, then add tape delay using a modified cartridge machine! But this procedure is hardly in keeping with the goal of producing high quality audio for the station's audience. Besides, products which can better solve the broadcaster's dilemma of how to provide a consistently high level of programming when switching among various sources are now available. Of course, some of the most successful applications incorporate noise reduction.

NOISE REDUCTION THEORY REVIEWED

The compander (compressor/expander) has been around for some years. Many attempts have been made to apply this concept to professional audio and broadcasting, but most of these have been less than successful for technical reasons.

The dbx approach uses a voltage-controlled amplifier as the control element and a true RMS circuit to sense the signal. These optimize the compander performance for ease of operation, maximum noise reduction, and increased headroom for the storage or transmission medium.

The noise reduction system consists of an encoder (with a constant 2:1 compression ratio) and a complementary decoder to precisely recreate the dynamic range of the original program material. The noisy medium (tape recorder, telephone line, microwave link, etc.) is sandwiched between the compressor and expander, with the dynamic range of the input signal cut in half during passage through the noisy channel.

Noise reduction is due, in part, to the masking effect of the program signal on the channel's noise level. For example, high-level musical signals may be loud enough to mask the perception of most noisy channels even without noise reduction. During quieter passages, however, the floor noise becomes audible. By placing the compressor before the noisy channel, the signal going through the channel is kept at high levels, so that more effective masking can take place. On playback, the expander restores the signal level to its proper value, while pushing down the noise at the same time. During periods of silence in the input, channel noise is pushed down so far as to become inaudible.

For a given signal-to-noise ratio, levels within the channels can be held down. The lower signal levels almost invariably result in less distortion of the signal. Because of the noise reduction system's action, the normal degradation of the signal-to-noise ratio with less signal will not take place. In addition, an increase in headroom is also realized.

THE dbx APPROACH TO COMPANDERS

For noise reduction to work in reality, a more sophisticated approach than has been suggested above is necessary. For a compander to operate properly, the compressor and expander must operate as mirror-images. To control the expansion/compression operation, each must have a signal detector.

There are three common types of signal detection: peak, average, and RMS. Since peak and average detectors are sensitive to phase shift, these types of signal detectors are inadequate for broadcast applications. To cope with time-dispersive media—channels with non-flat group delay, where relative phase shifts of different frequencies can cause severe alterations of wave shapes—dbx uses RMS level detectors in both compressor and expander. RMS detection measures the sum of the energies present in the signal and is not sensitive to phase shift introduced by time-dispersive media. RMS level is unaffected by irregular group delay. Peak and average level, however, are significantly altered. The use of RMS detectors insures much closer tracking between compressor and expander levels than would be possible with other detection schemes.

Since the dynamic range of live music sometimes approaches 120 dB, a means of automatically varying level in response to the detector's output over a 60 dB range must be used to achieve the 2:1 compression ratio. In addition, the dynamic range of the gain control element itself must be 120 dB or more, in order to avoid degradation of the input signal. To fulfill these criteria, dbx employs a voltage-controlled amplifier (VCA), which uses log-anti-log techniques. At the same time, the VCA control constant of −6 mV/dB, with 0 V control corresponding to 0 dB (unity) gain, is a fortunate match for the RMS detector's output constant of −6 mV/dB, with 0 V output corresponding to a presettable reference level.

Leslie B. Tyler is Chief Engineer of dbx, Inc., Newton, Mass.

To better cope with high frequency noise content within the channel, and to combat tape modulation noise, a high-frequency pre-emphasis of approximately 12 dB is used in the encoder, with a complimentary de-emphasis in the encoder. The de-emphasis serves to reduce high frequency noise by 12 dB when the signal is decoded.

Unfortunately, the 12 dB pre-emphasis alone would impose a headroom requirement on the transmission channel which may not be met in reality. Tape recorders which run at slower speeds are especially prone to high level, high frequency, saturation effects. To deal with this problem, dbx includes level detector pre-emphasis. This causes the RMS detector to weigh high frequencies more heavily than low frequencies in order to complement the signal pre-emphasis used and the anticipated overload situation.

TYPE I AND TYPE II NOISE REDUCTION

Since overload characteristics of tape machines vary drastically with their operating speeds, different amounts of level detecting pre-emphasis may be required. For example, the dbx Type I system is optimized for operation with high-speed tape machines (15 ips or greater), or transmission channels with wide bandwidths and generous high frequency headroom. Detector pre-emphasis begins at approximately 1.8 kHz, and boosts high frequencies by as much as 19 dB. The Type I detector bandpass is 20 Hz to 27 kHz.

The Type II system is appropriate when the bandwidth of the medium is narrow (particularly at the high frequency end), when peaky high frequency response is present, or when headroom is limited at high frequencies. Type II is therefore especially useful for 3¾ ips and 7½ ips tape machines, as well as telephone lines and other restricted media. Detector pre-emphasis in the Type II system begins at approximately 440 Hz, and reaches a maximum boost of 20 dB. The bandpass is 30 Hz to 10 kHz. When used properly, both systems can deliver in excess of 100 dB dynamic ranges.

BROADCAST NOISE REDUCTION

For the broadcaster, the primary limitations to signal-to-noise ratio lie in STLs (studio-to-transmitter links), telephone line feeds, cartridge tape machines, and video tape recorders. Although modern broadcast consoles have excellent signal-to-noise ratios, the best console in the world cannot show off its wide dynamic range when processed through a noisy STL.

Telephone lines are notoriously noisy, and when adequate bandwidth is available for proper encoder/decoder tracking, can greatly benefit from a noise reduction system. Furthermore, cartridge machines frequently suffer from headroom limitations and poor signal-to-noise ratios. Audio sections of video tape recorders are often relegated to second place in their overall design. For these situations (noise reduction around channels with limited high frequency headroom and response), dbx recommends Type II noise reduction.

When noise reduction is added to all station sound sources, the differences in dynamic range between these sources may be so small as to be inconsequential. A note of caution: when using any noise reduction, equipment such as equalizers, compressors, etc., must be kept out of the encode/decode loop. While the linear 2:1 compression/expansion ratios insure that level variations will not cause decode errors, any alteration of frequency response or dynamic range will introduce tracking errors.

SATELLITE COMMUNICATION SYSTEM APPLICATIONS

A compander system can be used to compress the audio signal sent around the world via satellite link. Unlike the dbx companding systems described earlier, the system designed for satellite communication system applications is a 3:1 compressor/expander system. It contains highly specialized pre-emphasis and de-emphasis networks and very tight frequency response and tracking specifications. The dbx 321 compander system was built on an OEM basis for National Public Radio System's ground station installation. It allows NPR to transmit high quality audio signals over otherwise very marginal satellite channels.

CUSTOM-TAILORING COMPRESSION RATIOS FOR DIFFERENT BROADCAST APPLICATIONS

Many areas in broadcasting require varying compression ratios suited to particular applications. As an obvious example, the classical music station might desire a gentle compression ratio for much of its programming as well as a higher ratio to improve intelligibility during news broadcasts.

Unlike compressors in the noise reduction systems, this compressor is a feed-forward device, so that very high compression ratios (limiting) can be achieved with absolute stability. The maximum compression ratio available is ∞:1, wherein no increase in output takes place once the input is above the compression threshold.

The level detector used is an RMS detector. It has been chosen here because of its unique time constants and the closeness of its response to that of the human ear. The time constants of RMS detectors are such that more weight is given to a large increase in level than to a small increase in level. This results in an attack time which varies with the amount of change in signal level, getting shorter for larger level changes. The attack time may vary from 15ms for a 10 dB change to a 3ms for 30 dB change. The release rate *(not* time) is fixed at 120 dB/sec (dbx model 160 specifications).

The compressor/limiter provides a gentle, natural sounding compression effect, keeping levels stable on an RMS basis, similar to the way in which the ear perceives levels. Since it will not over-react to peaks, the compressor/limiter should *not* be used as a device to prevent instantaneous over-modulation of a transmitter. However, it *can* act to control levels in a natural-sounding fashion, before the signal is sent to a final, peak-activated, limiter.

EXPANDING DYNAMIC RANGE

Although some form of compression is a broadcast fact of life, some broadcasters, seeking to enhance radio station sound, are experimenting with dynamic range *expanders.* Stations are using this equipment to reduce surface noise from records, emphasize detail in the nuances of the music, and generally capture a greater sense of the live performance. They are also finding that use of such signal processors can contribute to station "identity," which helps listeners recognize the station by its special sound character.

Splitting the audio frequency range into several bands brings freedom from undesireable side effects of expansion, since time constants are tailored to fit the bandpass of each section, and expansion of low frequencies will not cause pumping of high frequencies, and visa versa.

CONCLUSION

In today's highly competitive marketplace, increasing numbers of broadcasters are seeking high quality signal processors to custom tailor their station's sound to provide a marketing edge, to provide maximum talk power, or to clean up a noisy land line.

As with many companies, dbx engineers are cognizant of broadcasters' needs, and are interested in establishing an on-going dialogue on signal processing equipment and signal processing problems. ■

db Special Report

JOHN M. WORAM

What's New at the FCC?

EARLY THIS YEAR, optimistic Washington watchers thought they may have detected signs of life on 'M' Street, Northwest. At the headquarters of the Federal Communications Commission, it appeared as though the broadcast and recording industries were about to get an invigorating boost: In April, the Commission was expected to choose from among five competing AM stereo systems. At about the same time, the Commission was also going to resolve the matter of FM quadraphonic broadcasting.

Finally, it appeared that 1980 would be the year in which the FCC broke a long-hallowed tradition, and actually began to *help* the industry it serves.

Guess again! Perhaps in recognition of its position as the most inefficient federal agency within the US Government (according to the General Accounting Office), the Commission has once again managed to stifle technological advancement.

Commission-watchers speculate that AM stereo is now stalled until next year at least. FM quad is in no better shape, despite the fact that the matter has been before the Commission since *August, 1971!*

AM STEREO

Five AM stereo systems have been competing for FCC approval: Motorola, Magnavox, Harris, Belar, and Kahn. In April of this year, the Commission approved the Magnavox system, setting off industry-wide protests. (At least some of th. howls were predictable: with five systems in the running, *any* decision at all is guaranteed to displease 80 percent of the contestants.)

In the ensuing debate, it was learned that the FCC had prepared an "evaluation matrix," in which each system was graded on mono compatibility, interference characteristics, coverage, transmitter stereo performance and receiver stereo performance.

At first, the Commission denied a request (by Harris) to make the matrix public. Meanwhile, FCC staffers, preparing to justify the pro-Magnavox decision, acknowledged "ambiguities and omissions" in the record.

In its August 4 issue, Broadcasting Magazine published a "revised AM stereo evaluation matrix," presumably derived from FCC data. This shows Motorola as the high scorer, with 67 points (out of a possible 100). Belar is second, with 53, and Magnavox and Kahn, with 51 each, are tied for third. In fact, only Harris comes in lower than Magnavox, scoring 50 points. The Commission has not yet issued a satisfactory explanation for designating one of the third-place systems as the winner. However, it should be noted that the matrix does not give scores for coverage or receiver stereo performance to any system, due to insufficient data. Obviously, filling in the missing data could have a significant effect on the ratings. Did the FCC base its decision on the missing information? We don't know, but at the FCC, chief scientist Stephen Lukasik apologized for the "ambiguities, discrepancies and inconsistencies."

From the thoroughly muddled proceedings to date, one thing appears clear: the FCC is not in command of the situation. It seems to be incapable of making an intelligent evaluation, even if this might be (as Commissioner Quello inquires) that all five systems are minimally acceptable.

If we may indulge in a bit of over-simplification, it would seem that the Commission might make one of the following statements:

A. One system is clearly superior.
B. One system is *not* clearly superior.
C. We can't tell if A or B is correct.

On July 31, the Commission approved a Further Notice of Inquiry on AM stereo, which seems to indicate that C is the answer. That being the case, why did they approve Magnavox in April?

In engineering matters, one need not be embarrassed by requiring more data in order to reach a conclusion. However, when conclusions are published before the facts, we must come to some depressing conclusions of our own concerning the competency of our federally-appointed decision makers.

We suggest that the industry deserves better treatment from its regulators. When setting standards that will have a long-term effect on radio and recording, we don't expect the FCC to play "let's run it up on the flagpole, and see if anyone salutes." By a stroke of good (or maybe bad) luck, when the Commission ran Magnavox up on the pole, no one saluted, and the flag was taken down quickly, with apologies. That's really unfair to the winner, to the losers, to all of us. In fact, the "winner" becomes a loser too, and the FCC's already tarnished image gets another scratch on it.

FM QUAD

At first, it may seem unfair to blame the Commission for the apparent "death" of quadraphonic sound. Indeed, some Commission insiders cite the poor showing of quad in the marketplace as justification for taking no further action.

However, it is a fact of life that the record marketplace is profoundly influenced by broadcasting practice. It is also a fact that, over the past ten years, the Commission has allowed some quad systems on the air, while others were kept off. Therefore, the listening public has not been able to hear all the contenders in the quad marketplace. Based on what has been heard, the public has not come to the support of any quad broadcast system. And manufacturers, anticipating eventual action by the Commission, have become reluctant to invest in any of the competing systems, for fear that it may be rendered obsolete, once the Commission does make up its mind.

And so the matter stands. But at last, the Commission has finally acknowledged that even the discrete (4-4-4) systems are capable of operation within the present broadcasting system. So it would seem that a standard might be written that would allow such broadcasts to take place. (It should be noted that the Commission has acknowledged that matrix standards are unnecessary, since such programming is already possible under existing stereophonic standards.)

Well then, we can expect the Commission to finally do something! Right?

Wrong! The Commission wants yet another period of comments. After all these years, they now want to know if they should; A, specifiy a compatible discrete system, or B, leave the matter entirely up to the marketplace. The FCC calls the latter a "general standards" approach.

Well, "column B" certainly sounds like the good-old American way of doing business, doesn't it? It's a wonder it didn't occur to them ten years ago, when there was still a marketplace out there.

But "specific" or "general," what's the *real* difference between A and B? If a specific discrete system is standardized, the broadcaster may either use it, or not use it—the choice is up to each station, and its own unique marketplace. If a general standards approach is in effect, the broadcaster has exactly the same options. The significant difference is that this "no-standard standard" offers no frame of reference to either the broadcaster or the manufacturer. It needlessly muddies the waters, since the Commission has already determined that a discrete system based on the QSI/RCA proposals is preferable, and—unlike the AM stereo situation—there has been no significant opposition raised.

Of course, the Commission is no doubt sensitive to the trouble it brought on itself with AM stereo; a "non-decision" on FM quad will spare the Commissioners more discomfort. However, the industry has suffered through ten years of pain, countless millions of dollars in research, and man-years beyond counting have been wasted before the Commission. After all that, surely we deserve the courtesy of an intelligent response from the Commission. Just one, that is—it needn't become a habit.

Its been said before that the Federal Communications Commission is a little out of touch with the broadcast industry. In fact, the National Association of Broadcasters (NAB) has requested that a committee be set up between government and industry, to help facilitate a comprehensive approach to AM and FM matters now before the Commission. The FCC has considered the request, but has decided to postpone action for the time being. Enough said? ■

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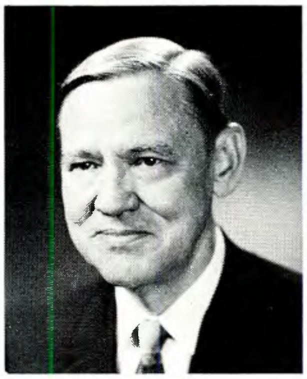
People/Places/Happenings

• **AVAB America** announces its move to new and larger quarters. The facilities are located at 967 Howard Street, San Francisco, CA and will be used both for sales and manufacturing. Among the products being built there is a new system of modular dimmers. The first complete permanent installation of this system is in the studios of **KOKH-TV** in Oklahoma City, OK. The dimmers are designed to complement AVAB's line of computer-controlled and manual lightboards. AVAB is also manufacturing the FM 800, a production sound mixer for the motion picture and television industries.

• **Audio-Technica U.S., Inc.** has named **Shane O'Neil** to the position of director of public relations. His primary responsibilities are for the planning and directing of all news media activities concerning Audio-Technica and its product lines. In addition, he will be involved in advertising and sales promotion projects. Prior to joining A-T, O'Neil had been director of public relations and merchandising for a Milwaukee advertising and public relations agency and held a similar position for five years with **Koss Corporation,** a high fidelity component manufacturer.

• **Quantum Audio Labs, Inc.,** a Los Angeles based professional console manufacturer, has acquired **Audio Logic**—a spin-off of **Uni-Sync, Inc.** Audio Logic's operation will be consolidated into the Quantum Audio Labs' factory facility in Glendale, California. According to **John Pritchett,** President of Quantum Audio Labs, Inc., sales and service for both product lines will be handled out of the Glendale facility—with Audio Logic operating as a wholly owned subsidiary of Quantum Audio Labs, Inc. **Andrew C. Thompson,** former president of Audio Logic, has been named National Sales Manager for Quantum Audio Labs, Inc. and Audio Logic.



John E. Volkmann

John E. Volkmann, a pioneer in the field of room acoustics and electroacoustics, died July 9, 1980 in Princeton, New Jersey, after a short illness. His professional career spanned more than 50 years.

Mr. Volkmann was born in Chicago, Illinois July 26, 1905. He received his BS (1927), MS (1928) and Professional Degree (1940) from the University of Illinois.

He worked continuously with **RCA** in the field of acoustics, specializing in the development and application of large scale auditorium loudspeakers and stereophonic sound systems, as well as a consultant on architectural, electronic and acoustic problems. He contributed to the solution of innumerable projects, including stereophonic sound systems for the **Radio City Music Hall,** the **Hollywood Bowl,** the recording acoustics for Walt Disney's **Fantasia,** custom loudspeakers for the **New York World's Fairs** of **1938** and **1964-65** and the **Jones Beach Marina Stadium.** He provided the acoustic design for the RCA Italiana's 364,000 cu. ft. Studio A—regarded as the largest and first ever built for the recording of full scale operas and symphonic orchestras. He was responsible for the development and design of the sound systems for the **John F. Kennedy Center** in Washington, D.C. He applied his pioneering concept of variable acoustics to the design of the **RCA Victor Recording Studios** in New York City.

For more than three decades, Mr. Volkmann was in charge of the advanced acoustic development and theater engineering activities at **RCA Camden,** New Jersey. In 1964 he transferred to the **RCA Laboratories,** Princeton, New Jersey as a member of the technical staff. He retired from RCA in 1970.

He has written numerous technical papers for scientific journals and holds several U.S. Patents.

In 1962 he received the RCA Achievement Award for "Advances in the Development of Architectural Acoustics" and in 1967 the John H. Potts Memorial Award from the **Audio Engineering Society** for "Elegant Application of Acoustic Principles in the Development of Large Scale Loudspeakers and Sound Systems." He was a member of honorary scientific fraternities, a Fellow of the **Acoustical Society of America,** the **Audio Engineering Society** and the **Society of Motion Picture and Television Engineers.**

Submitted by Harry F. Olson

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5	25	45	65	85
6	26	46	66	86
7	27	47	67	87
8	28	48	68	88
9	29	49	69	89
10	30	50	70	90
11	31	51	71	91
12	32	52	72	92
13	33	53	73	93
14	34	54	74	94
15	35	55	75	95
16	36	56	76	96
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