THE SOUND ENGINEERING MAGAZINE
MAY 1981
$1.95

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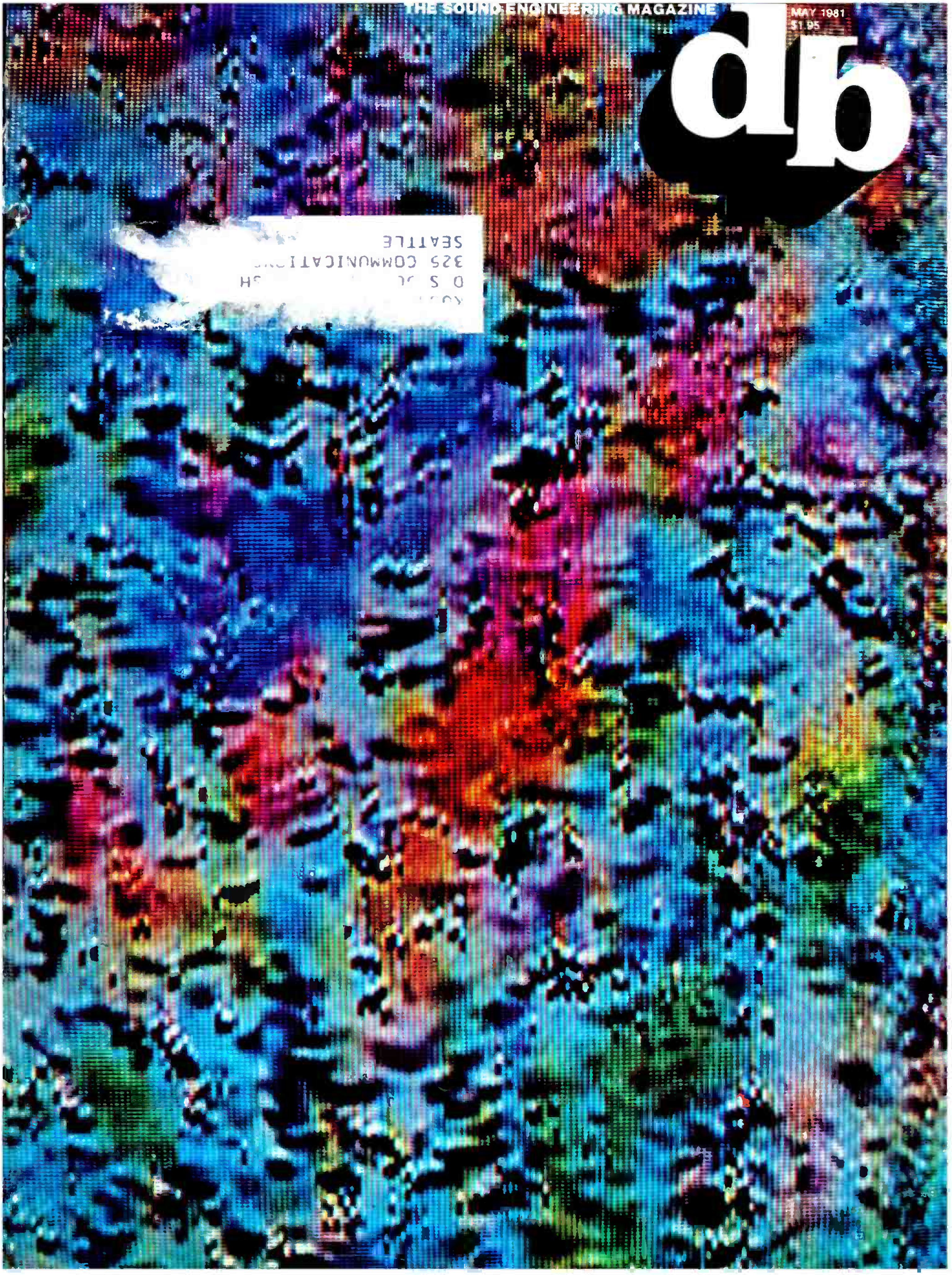

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SUBSCRIPTION NOTE

• This is the last month that we can offer the subscription rates found in this issue. Due to severe postal increases already instituted, we must raise the basic rate to $15.00 effective with all subscriptions received after June 1, 1981.

ABOUT THE COVER—

• This month's cover features a video animated photo from a laser video disc. Ron Hays is the video artist and Frank Serafine the production engineer of music and sound effects.

db

THE SOUND ENGINEERING MAGAZINE

MAY 1981 VOLUME 15, NUMBER 5

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db Letters

COLOMBIA'S WORLD-CLASS STUDIOS

To The Editor:

As the producer of some of the most successful records made in Colombia in the last few years, which have been issued all over Latin America, The United States and Europe, I want to take issue with the title of your article in the February 1981 issue called "Colombia's First World Class Studio." The implication that any studio not designed by Mr. Storyk is not "world class" has been highly resented and it simply is not true.

The fact is that in Colombia there have been for many years recording studios that represent the state of the art. Ingesón in Bogotá, Codiscos and Sonolux in Medellin, Felito in Barranquilla have been in operation for years. They have the latest multitrack equipment, with automated consoles (which the studio you describe does not have) and with acoustical design that has been highly praised by experts. In addition, those studios and others in this country have highly qualified engineers, and I submit that the human element and not the architectural design is what makes a recording studio "world class."

It is too bad that for one studio to start operation, it must denigrate the very good jobs done in the past in Colombia. Maybe they do not have enough confidence in their own capabilities.

Carlos Avellaneda

To The Editor:

I have read with interest in your February edition an article about "Colombia's First World Class Studio." As it seems that the 5000 miles of distance between Bogotá and Plainview doesn't allow for enough information, I would like to make something "perfectly clear" to quote one of your eminent men of state:

Referring to "first": We have been in existence for fifteen years.

Referring to "world class": I am not sure what that means, but regarding our installations I would like to inform you that:

a. We have a complex of seven studios, with excellent acoustic design, with 24 and 16 track MCI recorders; with 500 and 600 series MCI consoles with automation and with all the paraphernalia that a state of the art studio has. Their acoustic design has been praised by eminent authorities.

b. In addition, we have cutting, movie and video facilities.

Index of Advertisers

Coming Next Month

• In June, our featured subject is once again Microphones. We'll have an article on PZM theory and practice, and some more to say about what happens when you combine microphone polar patterns.

Also, have you been wondering about the audio in Billy Bob's Texas? This quaint little bistro in Fort Worth can handle some 7,000 visitors at once. We'll tell you all about it next month.

c. Important musicians and producers not only from Colombia, but from the United States, France, Aruba, all of South America and England have used and continue to use our studios.

d. We have rented and continue to rent to companies such as Philips-Polydor, CBS, RCA and others that would hesitate not to use a "world class" studio.

e. We have done commercial work for companies that you probably have heard of, like Coca-Cola, Marlboro, Pan Am and (without hyperbole) hundreds of others.

The only thing we have not done is to use the services of Mr. John Storyk, who by the evidence is as good an advertiser as an architect. We hope that a *sine qua non* condition to be "world class studio" is not to use Mr. Storyk's services. With this only exception, we believe that we qualify for the title.

MANUEL DREZNER
President
Ingeson Ltda.
Bogota, Colombia

db replies:

Recently, we've been trying to expand our international coverage by including more reports on studios around the world. However, due to the realities of time, geography, and the "bottom line," we are often prevented from personally visiting either the studio, or the city and country in which it is located. And so, we depend on our overseas readers to share their work with us, and with our other subscribers.

As for Mr. Storyk, he's not that much of an advertiser. (By the way, has it occurred to anyone that the designers, builders and most of the console manufacturers mentioned in the February issue have never *advertised in* **db**?) *What Mr. Storyk* does *have going for him is a damned good PR department, which supplies us with frequent reports of his projects. In fact, there's another one in this very issue. In a burst of innocent (?) enthusiasm, Fonovision Internacional was described as Colombia's* "first" *world-class studio. Now that we've heard from Srs. Avellaneda and Drezner, we realize that "first" should have read "newest."*

In the future, we'll be more attentive to qualifying adjectives like "first," "world-class" and such. But you can ease the burden on us, too—don't be shy! Let us hear about your studio projects. There are so many of you, and only one of us. So don't wait for us to find you; let us know you're out there. We can both profit from that sort of communication.

DEPT. OF MYTH-INFORMATION

TO THE EDITOR:

I recently received the February issue of **db** and was attracted immediately to your article about RCA's studio in Mexico City because I have made several recordings in that city. Although I have not worked in the new studios, I have utilized the personnel and equipment of RCA. In fact, my first recording there was made in collaboration with Mario Sanchez Roldan.

Of course, I was interested to read his comments about the new construction. However, I was a little shocked to hear him quote—and, I must say, to watch you endorse—a very strange theory of acoustics. Sr. Sanchez Roldan continually refers to the "fact" that sound travels faster in rarefied atmospheric conditions than at sea level. This is simply not true—as a glance at any acoustics text will reveal. The speed of sound in air is somewhat dependent on temperature, but in no way dependent on air pressure.

As so much of RCA's studio construction seems to have been devoted to counteracting this non-existent effect, it makes one wonder about other allocations of their time and money. Perhaps the clue to the whole thing was contained in the fact that they actually built little pyramids into the ceiling to take advantage of their metaphysical powers. This corroborates the theory that the Egyptians were well ahead of their time by constructing their famous basement recording studios.

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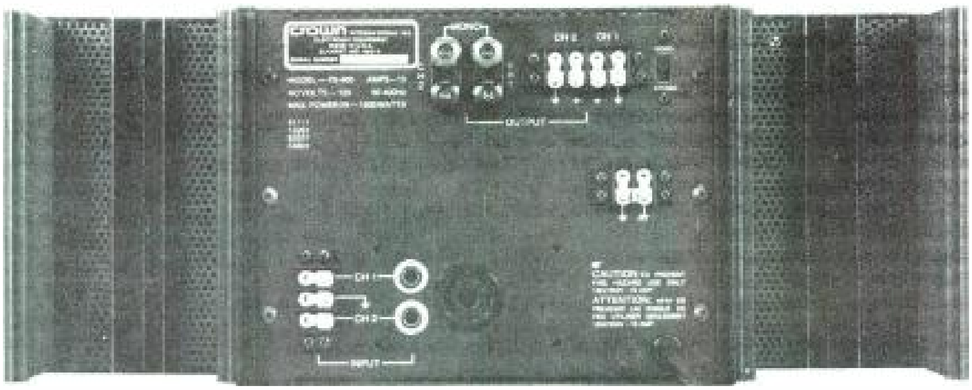

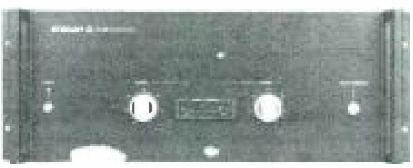

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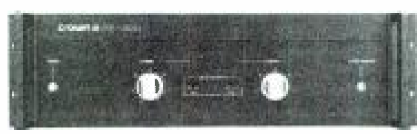

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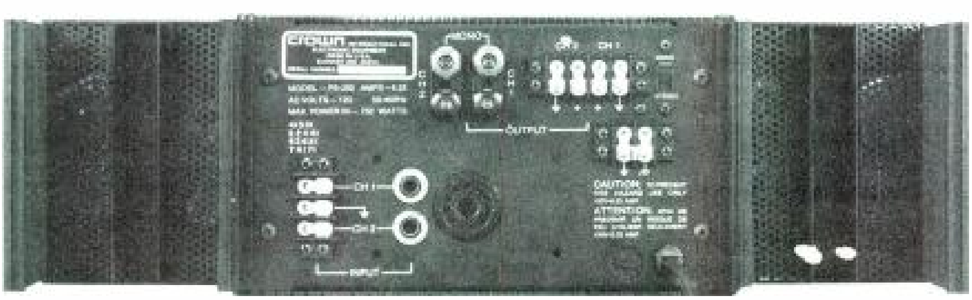

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In addition to all of this, I have not been able to figure out what is meant by the "narrower dispersion of sound both vertically and horizontally." What sound? Any sound? The sound of a Bass Drum? The sound of a French Horn?

It makes me a little nervous when a magazine devoted to Sound Engineering starts publishing articles containing Sound Myth-information.

ANDREW KAZDIN

db replies:

As Mr. Kazdin correctly points out, the speed of sound does not vary with air pressure, although it took us more than "a glance" to verify this. Some texts say nothing about air pressure, while others imply a dependent relationship by noting the barometric pressure at which the measurement was made.

Finally, we turned to Harry Olson, who writes, "...there is no change in velocity due to a change in pressure." Music, Physics and Engineering, *Dover Publications, 1967.) In another reference, "...the speed of sound in a gas is independent of pressure."* (Elements of Physics, *Prentice-Hall, 1955.)*

Unfortunately for us, we forgot most of our physics theory immediately after escaping high school, and so this point slipped right past us (no doubt travelling faster than the speed of sound). We've dispatched a letter of inquiry to Mexico City for clarification of the design parameters under discussion there, and hope to have more information later.

In the meantime, we repeat—"one more time!"*—that the printing of a person's views here is not an endorsement of them. In the past, we have printed the opinions of a variety of acousticians, most of whom have one thing in common: they disagree with each other. In this context, we hope that* **db** *will continue to be a forum for the often conflicting opinions and theories of studio designers.*

AUDIO ON VIDEOTAPE

TO THE EDITOR:

In the January '81 issue your editorial mentioned (2nd paragraph) that VHS or BETA format video tape recorders are being used for two-channel recording or mixdown. How? Perhaps an article on this subject would be appropriate. I know that the Sony U-Matic machine already has 2 audio tracks (of rather low quality for critical uses) and the BETA format home unit has one audio track (not up to the quality of those on the U-Matic). So how do they get professional quality recordings from this equipment? Please explain.

R. DENNIS ALEXANDER

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db replies:

The article is not only appropriate, it's in this issue of **db**! *In "The Sony Digital Picture," Curtis Chan gives us a detailed explanation of the way in which high-quality audio is stored on videotape. As you'll soon see, it's quite different from the audio you describe in your letter.*

BREEDS SYNTHESIZERS, NOT HORSES

To The Editor:

HEY! Ralph Hodges is careless with the facts! Contrary to what he "reports" in your March issue, I am not breeding horses in New England. I am designing synthesizers in western North Carolina. Custom synthesizers. ANALOG custom synthesizers.

I would appreciate your putting this in your letters column so people don't trek through the horse farms of Connecticut looking for me.

Robert Moog
President
Big Briar, Inc.
Leicester, N.C. 28748
(704) 683-9085

db replies:

Would you buy a used horse from anyone named Moog? Probably not, but if you're looking for an analog custom synthesizer, he's certainly the horse's mouth. Get in touch with him in NC, not CT.

db Calendar

MAY

19-21 **Syn-Aud-Con Audio Industry Seminar.** San Juan Capistrano, CA. For more information contact: Syn-Aud-Con, P.O. Box 1115, San Juan Capistrano, CA 92693. Tel: (714) 496-9599.

JUNE

10-12 **APRS '81; 14th Exhibition of Professional Recording Equipment.** Kensington Exhibition Centre, London, England. For more information contact: E. L. Masek, 23 Chestnut Ave., Chorleywood, Hertz, U.K.

21-7/3 **Audio Workshop.** Concordia College, Moorhead, MN. For more information contact: Audio Workshop, Concordia College, Moorhead, MN 56560. Tel: (218) 299-4201.

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MARTIN DICKSTEIN

db Sound With Images

The State of the Art

• Everyone knows that muti-image, multi-media, multi-screen presentations are here to stay. This, of course, does not mean that everybody is getting into a lot of slide projectors mixed with film projectors on a lot of screens. It does mean that lots of projects are involved with this sort of display for many various purposes and applications. And lots of money and time are going into these events.

THE NUMBERS GAME

Let us take a look at some of the numbers that may boggle the mind as much as the flashing images on the screen. Last year, at the end of October, a multi-image festival took place at which more than 1,000 people watched a week of multi-presentations. To present the 174 shows which vied for awards, about 64,000 slides were used in almost 1300 trays. There were 93 projectors set up in a variety of screen formats, with 30 programmers from different manufacturers, 100 dissolve units, and 150 lenses. Add to this 16 loudspeakers, 10 amplifiers, tape recorders, equalizers, etc., and you have an armada of equipment which took about 2,000 person-hours to set up and operate. Does the brain see flashing $$$$$$? Consider that this is only a small part of what goes on all year for sales meetings, conferences, in-house presentations, new product introductions, educational and government institutions, and a host more.

A couple of years ago, a report came out which issued some more mind-boggling facts. In 1978, over 250 million original slides were produced (about one-fifth more than the previous year). Slide production cost over a billion dollars, with total expenditures for the slide industry running over 1¼ billion. (These figures are up over 10% from 1977.) Almost 3,000 companies are involved, one way or another, in the slide business. If the trend continues, and it is expected that it will, do you dare guess where the numbers will be at the end of this and next year?

Let's take a look at another event that took place earlier in the year. This was an international multi-affair in which 118 entries showed 127 presentations. It was estimated that almost $40,000 worth of equipment was used, including 5,000 feet of cable, 150 feet of screen, 130 projectors, 6 manufactured hardware systems, several hybrid systems, a cherrypicker, etc., etc.

A TRIP FULL CIRCLE

How did we get here and where are we going? And how does the recording engineer fit into all this? It all started, of course, with the development of slide film. Once there were slides, there had to be a projector to show the slides. After "clunking" away for a while, single projector—single slide at a time, there came multi-screen (2 screens with side-by-side images), dissolvers with 2 projectors on one screen eliminating the black space between individual slides, then dissolvers for 3 projectors on one screen, 3 dissolvers for 9 projectors on 3 screens, then programmers to move the slides faster, then 5 dissolvers for 15 projectors on 3 screens with 2 overlaps, and now the computer to make slides move so fast with all sorts of special effects that we're getting motion and visuals which rival film and video.

Now, it seems we've come full circle. As long as we can rival film, why not work with all the projectors we need on one screen? A great many presentations are now being made for just that screen format—single screen, with several-to-many projectors, computer programs, and many special effects. In fact, at one of the multi-meetings where all the projectors we mentioned earlier were used, 17 of the 27 finalists for awards were in single-screen format. Some of these, of course, included 2 and 3 projector arrangements, but many were for more projectors.

In selecting the finalists, the subjective reaction of each of the judges to each presentation played a part in the final choices. In addition, each of the judges had to be concerned with script, visuals, audio, programming, concept, and achievement of stated objectives by each of the producers. Did you happen to

notice that audio was one of the criteria used to choose the winners? That included the quality of the sound, the balance of the mixes, the relative levels of voice to music, sound effects to voice and music, etc. It also included the quality of the reproduction and distribution in the presentation arena. It also meant the choice of music or effects in relation to the visuals. Not all of the music had to be hard rock with a fast beat so the slides could move fast. It meant the aptness of the music for the concept and effects in relation to the show. In all of these areas concerned with the audio part of the audio-visual display, the recording engineer can play an important role.

THE AUDIO PORTION OF THE PRESENTATION

Consider some of the expenses involved in the audio portion of the presentation. With all the money that is spent on creative effort, photography, art work, duplication, programming, staging, etc., some estimates show that the audio part runs roughly 15 percent of the total production. For example, there's the professional announcer (maybe union) who can get from $100 to $400 for the first hour, and somewhat less for additional time, plus the percentages to the union and the agent. There's the producer who can get somewhere around $50 an hour or between $300 and $600 per day. There's the music library from which music is rented at a per-cut fee, or a lump sum (something around $200) for a 10-minute total, including clearance rights. There's the writer of original music and the scoring and orchestration, which can range anywhere, depending on the fame of the composer, to perhaps $10,000 for about 5 minutes, not including the studio and musicians. Add to all this, the time to assemble the parts, mix down time to stereo or mono (from 8 or 16 tracks), and there's an additional cost of maybe $60 to $160 per hour. (Please remember that all of these costs are only estimates and just reflect relative comparisons, not the actual prices which depend on geographic location, union requirements, etc., etc., etc.)

With all these multi-media shows taking place in so many hotels, board rooms, conference centers, theaters, concert halls, and almost anywhere else they can put up a screen and projection equipment, the recording engineer has very little control over where and how his fine editing and mixing work will be presented. The usual is for the work to be done according to the instructions of the producer. Remember, however, that it is also usual for the production to start with the soundtrack. Almost all of the time, with only a few exceptions, the producer has only an idea on what visuals will be presented, and the sequence of images are roughed in story-board form. But the actual slides and effects have not yet been put into the drums, nor does any of the programming take place until the sound track is finished. Hundreds (sometimes thousands) of slides are conceived or shot, but their actual positioning cannot take place until the track is completed so their movement can follow the track. So, now, the sound track becomes the critical part of the show, and all else follows. The music has been written especially for the show, or bought from libraries, to fit into the concept of the presentation. The visuals will be selected to work with the track. The announcer has been hired for his voice, and reading of the script. It now becomes essential for the mix to be right.

Remember that the editing and mixing engineers are listening on professional equipment, through excellent speakers, in an ideal atmosphere for personal or small group listening. But the final show may not be blessed in the same way. The playback may or may not be on professional equipment through excellent speakers, and may also have to fill a hall with anywhere from 50 to 2,000 (or more) people. The acoustics will not necessarily be ideal, and each of the people in the audience, having a carefully selected hi-fi at home in the living room, professes to be a sound quality expert. The mix, therefore, becomes even more important.

The final tape should, of course, be of the highest quality possible. The end result should be as few generations away from the original tapes as possible. The balance of the voice with the music should be such that it not only sounds good in the mix studio but will sound great in the presentation hall. The relative level of the narrator over the music, depending on the aural quality of the announcer's voice and the type of background music, should be accentuated so that the audience will hear and understand every word. (After all, the producer already knows the words of the script practically by heart, and the engineer has heard the announcer's words without any background sound, so the blend may be recognizable by both, but the audience has not had these privileges, and must get the message of the reproduced mix the first time around—there's no second shot.) The engineer can be very valuable, with his expertly trained ear, in making the sound track as good, if not better, than the visuals. Remember, the mood of the show is in the music, but the message is in the voice.

THE VOICE

The use of an echo effect on the voice is not recommended. Except for special reasons, the voice should be as clean over music and sound effects as possible. Stereo sound is used very frequently with multi-image shows. It seems that producers like the idea of stereo sound with their jumping images. This may be well if the presentation room allows for this type of sound distribution. However, some areas are much wider than they are long. The screens are very wide, and the speakers may have to be placed far apart. The audience may spread beyond the screen, and some will hear only what is coming from the speaker on their side of the room. Stereo at home is one thing, but in a very large auditorium or ballroom, only the people in the center area back from the screen will actually get the full stereo effect. Sure, it's even OK if the different voices appear on separate tracks to follow the associated images on the screen, but even with this arrangement, the people on the side of the room opposite the speaking image may lose some of what is being said, and not get the full message for which the whole presentation was conceived in the first place. The recording and mixing engineers can again prove their worth when they find out they are working on a track to be used with a multi-image show, if they caution the producer against complete separation of voices. Perhaps the voice can be recorded full level on one track, and half-level on the other, so no part of the audience gets all or nothing. The illusion that the image on one side of the screen is speaking will still be maintained and nothing will get lost.

The traveling sound effect can be very effective. This takes at least a couple of tracks on the tape, and a smooth pan from one side to the other. Here, again, the brilliance of the pan will show, but if there happens to be a voice over the sound, be sure the words are not lost in the peak levels of the effect. It is still more necessary that the message come through rather than amazing or amusing the audience. Perhaps you can amaze them by letting them hear the voice clearly, or amuse them by making the effect in the clear without voice, but it is something to look out for.

With all the multi-type shows that have been produced in the last few years, almost all recording engineers have probably had the opportunity to see one or more. Abrupt slide changes usually follow the beat of the music, while dissolves come during smoother sections. Sometimes, a sudden drum beat can indicate a quick change on the screen. If the recording or mix engineer sees, or hears, the possibility of anything like this happening, it might be well to ask the producer if the sound, such as the drum beat, should be accentuated in the balance for greater effect. Sometimes, the engineer knows better what can be done with the equipment than the producer knows what to ask for. Help the producers in any way you can. They'll get better results, the clients will be more pleased, the producers will get more work, and so will you.

WHERE WE STAND

Where are we going with all this multimedia? In some shows, we will see film added more and more. In other areas, more projectors will produce "film-like" shows. In a great many instances, the swing will be back to less screens and projectors and more ingenuity and creativity...brilliance with simplicity. But, soon, there will be more lust for lasers. They will create fantastic images, also programmed to go with slides and film. There will be the development of 3-D images which will become part of the multi-image scene. And don't forget holography. They will soon work out ways this can be incorporated into shows as another programmable medium. Musically, there may be greater use of electronic music to go with electronic images.

Presently, we're using thematic music, or pictorial music, to enhance the visual images. We recognize the pictures on the screen and follow the music with our ears. Abstract images allow almost anything to happen aurally, and the more abstract the concept of the image, the farther out the music can go. In any event, the recording and mixing of the final track will still be the beginning of the show. Even if the visuals start to come first, the audio track will still have its very important part to play in the final show. Only by having the sound as good as the picture, can the term audio-visual really mean both halves are equal. ■

The Floating Point System

• Much of our previous discussion has implied that digital audio systems should always be of the highest quality, since they cost so much more than their analog counterparts. There are, however, some exceptions. If we examine the history of digital audio, we see that the first systems to be introduced were delay lines and reverberation systems. Both of these can be considered as secondary channel systems. Still another example of secondary channel applications is the special effect systems. Only the digital tape recorder, which is just now appearing, is a primary channel device. Systems—digital or otherwise—which are to be used in secondary channel applications do not require the highest quality. For one thing, their outputs are usually added to the primary channel at a reduced amplitude. The motivation to consider reduced quality is simply a matter of cost.

As a signal processing device, the digital delay line is particularly interesting since tape delay notwithstanding—it has essentially no analog counterpart. We should thus expect to find a complete quality-price spectrum. When we examine the digital delay line we observe that basically there are only two components to it: the digital memory element, and the conversion circuitry. The former is not so much a matter of variable quality, as it is just the amount of memory that needs to be purchased. Hence, the quality-cost relationship is determined almost exclusively by the conversion systems. Many of these use a simpler A/D rather than the straight 16 bit PCM which we have been describing.

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FLOATING POINT

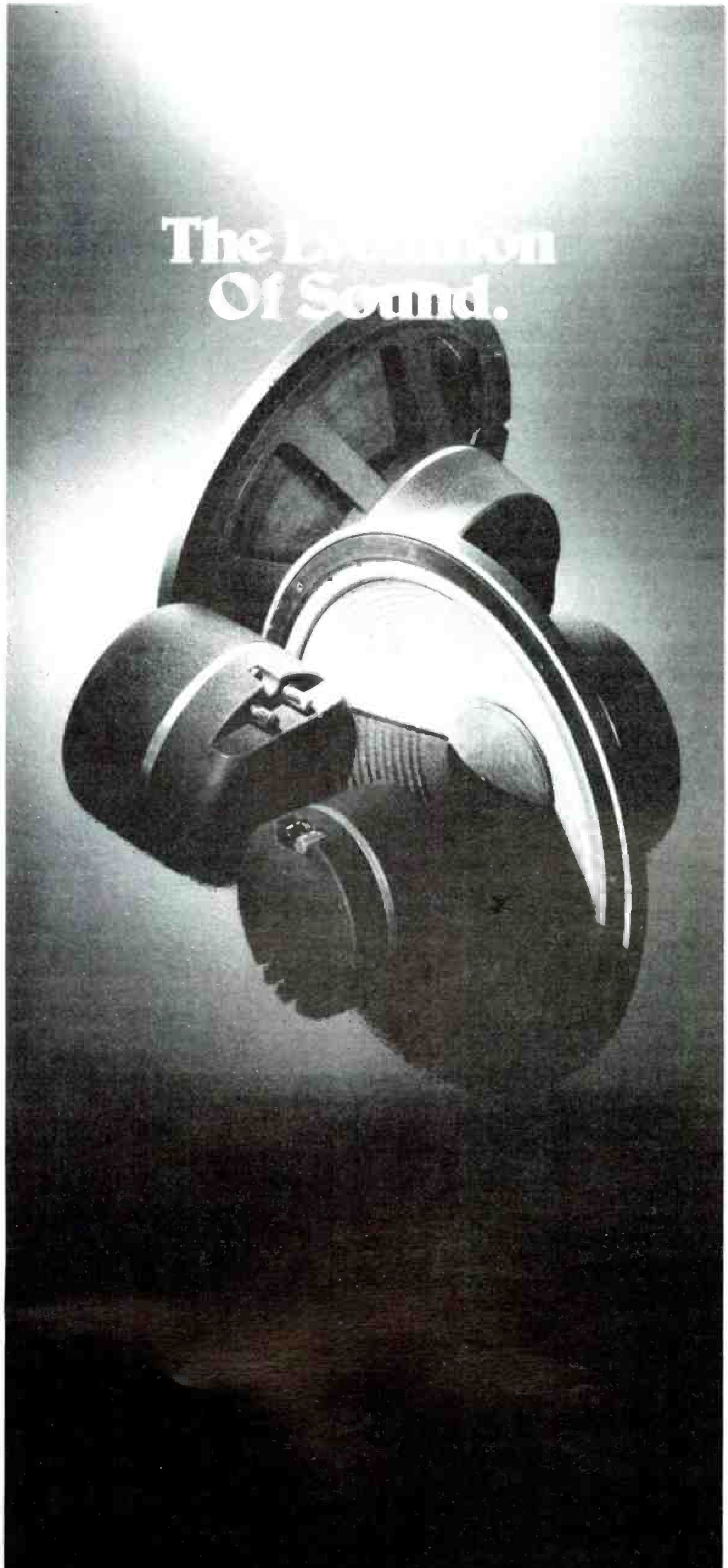
The most common A/D system in the economical equipment is called floating point. This name comes from the mathematical concept of the decimal point. Consider a calculator display which has six digits. This allows us to create numbers such as 10000.2, 2.34543, 435.989, etc. Notice that the resolution or precision of the least-significant digit is a function of the size of the total number. The number 10000.2 has a resolution of one tenth. We cannot add 0.01 because it would create a seventh digit. On the other hand, 435.989 has a resolution of one thousandth and we can easily add 0.01 to it. Another way of viewing these numbers is that the resolution is a percentage of the actual number. The definition of this type of number is a floating point with a fixed number of digits. Because the decimal point can move, or "float," the range of the numbers is from 999999.9 to .9999999. Notice that the local accuracy is smaller than the range.

For an audio analogy, think of the fixed number of digits as being the number of resolution bits. Now, we'll provide a second piece of information to specify the location of the decimal point. Consider the case of a 10-bit digital word. It can range from +511 to −512 and it can take on any value (integer) in between. Suppose we add two extra bits, whose meaning is the control of the decimal point. This auxiliary exponent word has values of 0, 1, 2, 3. Let us assign the following definitions to these numbers: the 0 means multiply the original 10 bit digital word by 1; the 1 means multiply the word by 2, the 2 means multiply the word by 4; the 3 means multiply the word by 8.

Now we can represent the number system as EE-M where EE is our 2-bit exponent, and M is the 10-bit mantissa. (These terms are taken from logarithmic notation.) The number 3-512 thus represents 8 × 512, since the 3 exponent means multiply by 8. The maximum range is now 4088 to −4096, but the smallest step for this case is 8. We have expanded the maximum signal capacity by reducing the resolution. But we can also represent numbers with a fine resolution when the exponent is small. The number 0-10 is just one unit greater than 0-9. We have just created a system which allows us to eat our cake and have it too. For large numbers, we use a coarse quantization; for small numbers, we use a fine quantization. Notice that a straight PCM representation which could span the range from +4095 to −4096 would require 13 bits and of course, it would require a 13-bit converter. The floating point representation of E = 2 and M= 10 is only 10 bits, and it requires a 10-bit converter with a switchable gain amplifier.

To further demonstrate the power of this, consider a 10-bit mantissa and 3 bits of exponent. This gives as a multiplicative scale range of 1 to 128, since that one extra exponent bit gives us four more states for the scaling. The maximum signals are now 65,408 to −65,536. The 13 bits can give us the range of a 17-bit system; however, large range and small resolution are *not* available at the same time. Although we can represent a number such as 65,408, the next smallest number is 65,280. With a straight 17-bit PCM system, the next smallest number would be 65,407. For small signals, when the exponent is 000 (gain = 1), the steps are still 1 unit apart.

In the table below, we illustrate the nature of the quantization levels for the floating point system. The left-hand column shows the number of the quantization level using floating point notation; the next column shows the same level using standard notation, and the third column gives the difference between neighboring levels. An input signal is quantized in the usual fashion, and each voltage sample is assigned to the nearest quantization level. Thus, 510.2 is assigned the level 510. The value 513.1 is assigned the level 514.



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Floating Point	Standard	Difference
.		
.		
.		
0*509	509	1
0*510	510	1
0*511	511	1
1*256	512	1
1*257	514	2
1*258	516	2
.		
.		
.		
1*509	1018	2
1*510	1020	2
1*511	1022	2
2*256	1024	2
2*257	1028	4
2*258	1032	4
.		
.		
.		

There are several interesting properties of this table. Notice that as the signal increases, the quantization error increases, thereby creating more quantization noise. This is a signal dependent noise. The second thing to notice is that certain quantization levels have two or more floating point numbers associated with them, although this is not apparent from the table. The level 256 could be created from the floating point number 0*256, but it could also be created from the floating point number 1*128 or 2*64 or 3*32 or 4*16 or 5*8, etc. This apparent inefficiency is actually used in the implementation.

SIGNAL-TO-NOISE

The performance which one will realize from such a system is very similar to that of an analog compander such as Dolby or dbx. These systems increase the noise level when the signal becomes larger. These systems do, in one sense, improve the performance in terms of dynamic range, but not in terms of signal-to-noise. To extend this discussion, let us make a careful definition of these two quantities.

Dynamic Range is determined by the ratio, or dB difference, between the maximum sinewave signal at a given distortion level and the noise observed when no signal is present. *Signal-to-noise* is determined in the same way, except that the noise is measured while the signal is present. The dynamic range is easier to measure, since an ordinary voltmeter can be used to measure both the signal and the noise. Signal-to-noise requires the measurement of the noise while the signal is present. A notch filter must thus be used to remove the signal from the measurement of the noise. This is not strictly correct, since the distortion harmonics of the signal will be included in the noise measurement. A better and more complicated method would be to use a comb filter to create a series of notches at the fundamental and all of the harmonics.

MASKING

In digital hardware, a floating point system is often acceptable because the increased noise is masked by the higher level signal. This same property is present in an analog tape recorder which may have 75 dB of dynamic range but only 40 dB of signal-to-noise. This comes from the well-known modulation noise. The relative inaudibility of the noise can be explained by the fact that the noise has a spectrum which is concentrated near the signal frequency.

With the floating point system, this is not the case. The quantization error generally creates a noise spectrum which is white. We discussed this in previous articles. The white spectrum means that the noise has equal energy at all frequencies. If the program is also broadband, then its overall loudness will mask the noise. However, one dramatic case which does not work well is for a low-frequency organ note. In this situation, the high amplitude of the narrow-band program forces the exponent to a high value, and the broadband quantization noise is quite large. We hear the noise increase without any masking effect.

Electro-Voice's Greg Silsby talks about the Sentry 100 studio monitor

Production Studio. WRBR-FM. South Bend, Indiana.

In all the years I spent in broadcast and related studio production work, my greatest frustration was the fact that no manufacturer of loudspeaker systems seemed to know or care enough about the real needs of broadcasters to design a sensible monitor speaker system that was also sensibly priced.

Moving to the other side of the console presented a unique opportunity to change that and E-V was more than willing to listen. When I first described to Electro-Voice engineers what I knew the Sentry 100 had to be, I felt like the proverbial "kid in a candy store." I told them that size was critical. Because working space in the broadcast environment is often limited, the Sentry 100 had to fit in a standard 19" rack, and it had to fit *from the front, not the back*. However, the mounting hardware had to be a separate item so that broadcasters who don't want to rack mount it won't have to pay for the mounting.

The Sentry 100 also had to be very efficient as well as very accurate. It had to be designed so it could be driven to sound pressure levels a rock'n roll D.J. could be happy with by the low output available from a console's internal monitor amplifier.

In the next breath I told them the Sentry 100 had to have a tweeter that wouldn't go up in smoke the first time someone accidentally shifted into fast forward with the tape heads engaged and the monitor amp on. This meant high-frequency power handling capability on the order of five times that of conventional high frequency drivers.

Not only did it have to have a 3-dB-down point of 45 Hz, but the Sentry 100's response had to extend to 18,000 Hz with no more than a 3-dB variation.

And, since it's just not practical in the real world for the engineer to be directly on-axis of the tweeter, the Sentry 100 must have a uniform polar response. The engineer has to be able to hear exactly the same sound 30° off-axis as he does directly in front of the system.

Since I still had the floor, I decided to go all out and cover the nuisance items and other minor requirements that, when added together, amounted to a major improvement in functional monitor design. I wanted the Sentry 100 equipped with a high-frequency control that offered boost as well as cut, and it had to be mounted on the front of the loudspeaker where it not only could be seen but was accessible with the grille on or off.

I also didn't feel broadcasters should have to pay for form at the expense of function, so the walnut hi-fi cabinet was out. The Sentry 100 had to be attractive, but another furniture-styled cabinet with a fancy polyester or die-cut foam grille wasn't the answer to the broadcast industry's real needs.

And for a close I told E-V's engineers that a studio had to be able to purchase the Sentry 100 for essentially the same money as the current best-selling monitor system.

That was well over a year ago. Since that time I've spent many months listening critically to a parade of darn good prototypes, shaking my head and watching some of the world's best speaker engineers disappear back into the lab to tweak and tune. And, I spent a lot of time on airplanes heading for places like Los Angeles, Grand Rapids, Charlotte and New York City with black boxes under my arm testing our designs on the ears of broadcast engineers.

The year was both frustrating yet enjoyable, not just for me but for Ray Newman and the other E-V engineers who were working on this project. At this year's NAB show it all turned out to be worth it. The Sentry 100's official rollout was universally accepted, and the pair of Sentry 100's at the Electro-Voice booth was complemented by another 20 Sentry 100's used by other manufacturers exhibiting their own products at the show.

What it all boiled down to when I first started the project was that I knew that the Sentry 100's most important characteristic had to be *sonic integrity*. I knew that if I wasn't happy, you wouldn't be happy. I'm happy.

Greg Silsby

Market Development Manager,
Professional Markets

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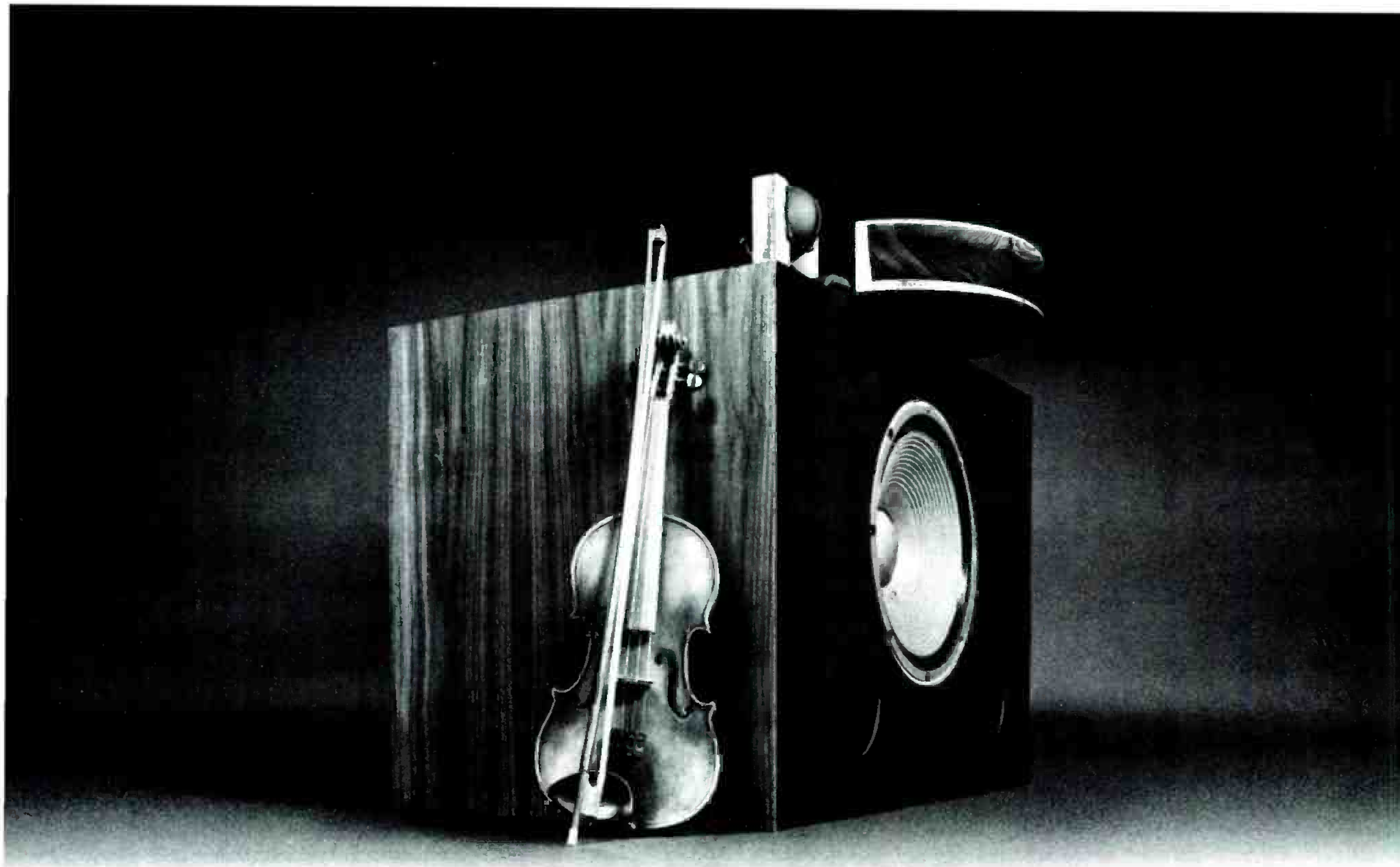
For most complex audio programs, this is not really a serious issue. However, one can easily try an experiment with a 20 Hz tone burst. Listening at the output, we will hear the burst plus a burst of white noise. In theory, the same experiment can also be tried at very high frequencies. A 10 kHz burst also does not produce much masking. Some manufacturers improve on the situation by using some amount of pre- and de-emphasis to minimize the effect. Almost all of the commercially available secondary channel digital audio systems use this kind of encoding. The basis is strictly economics, since a 90 dB dynamic range can be achieved with 10 or 12 bit A/D converters; the converter corresponds to the mantissa, not the exponent.

Another way of looking at the issue is to think in terms of a compander with the compression and expansion built into the same A/D conversion system. Floating point is thus equivalent to companding with all the advantages and disadvantages. But floating point companding is, in some sense, primitive, since the control is only on peak level and broadband. More sophisticated companders break the signal into spectral bands or use some kind of average/RMS to control the gain. One 50 μsec. peak in the signal forces the compander to go to the high quantization noise state. For other reasons not yet discussed, the return to the low quantization noise state is usually inhibited for something on the order of 100 msec. This will be discussed in the next article in which we shall deal with the implementation of floating point systems. We must remain aware of these issues to properly evaluate these systems.

If you are unhappy with all of these "defects," you can solve the problems with a mere application of $$$$. In the limit, the choice is always the user's. I personally feel that for secondary channel applications the defects are acceptable. It is interesting to note, however, that even the more expensive secondary channel digital systems are no longer using floating point representations. The cost of a straight 14-bit system has become low enough to drop the floating point. These 14 bit systems have much better signal-to-noise ratio, but worse dynamic range. ■



NORMAN H. CROWHURST

db Theory & Practice

• When my wife and I first arrived in this country in 1953, I was retained by Sherman Fairchild, who owned many companies, including Fairchild Recording. I acted as a technological consultant for him, as well as being employed by Fairchild Recording. One of Sherman's sayings was, "I have found in life that, if you are doing something right, you usually end up making a little money at it." His being a multimillionaire at a time when a million dollars was worth a lot more than it is today, testified that the principle must have worked, at least for him.

Over a decade later, I started writing this column, the purpose of which was to reconcile theory and practice, to show that both exist in the same world, and that there is never one answer that is right in theory, and a completely different answer that is right in practice. In retrospect, I realize that "doing it right" has a lot to do with reconciling theory and practice.

Let me briefly take an experience of my own that illustrates this. Back in the '40s, when I was in my 30s, I "contracted" conjunctivitis, which produced some very painful inflammation of my right eye. The whole eye was blood-red and after two weeks I had lost my vision in it. The medical doctor to whom I went (under the British National Health scheme) then made an appointment for me to have the "infected" eye removed, in case it might infect the other eye. He'd tried all the miracle drugs, to no avail.

That was when I decided I needed a second opinion, preferably not from a medical doctor. I went to a chiropractor. He didn't know if he could help me, but he gave me a manipulation anyway. Within minutes, the sight was back in that eye and by the next day, when I was due to go into the hospital to have the eye removed, it was back to normal, maybe slightly bloodshot.

An interesting sidelight of that incident surfaced over a year later. I returned to work with a certificate from the chiropractor that satisfied the boss that I was okay for work. But the National Health scheme wouldn't accept a chiropractor's certificate, and the medical doctor wouldn't give me one: he didn't believe in miracles! So the National Health officer, whose job it was to terminate my sickness benefit payments, told me he had a terrific run-around over that with his bureaucracy.

I've thought about that many times. Many, because of similar incidents, swear by chiropractors and condemn medical doctors. But really, it was a case of doing it right. My eye was not infected, but was reacting to a pinched nerve in my spinal cord. Putting that right corrected the whole situation. Had that not been what was wrong, the chiropractor's treatment obviously would not have worked.

In audio, a good example of doing it right occurs in the realm of curing obstinate hum problems. In the case of my eye, who would have thought that a maladjustment of my spine could cause loss of sight? Or that correcting the maladjustment would restore sight so quickly? We can remember many instances where a system hum came and went, when some thermostatically controlled equipment, such as a refrigerator, clicked in and out.

The hum would go up, say, when the refrigerator pump clicked on, and go down when it clicked off. Well, that could

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be induced from radiation put out by the pump motor, you'd say. But then a little later, you'd find that the situation was reversed: the hum went up when the motor clicked off, and went down when it clicked on again. The first time we encountered this kind of situation, we thought it was weird, very weird. What connection could there be?

Obviously there was one, but explain it. That was not so easy. In theory, there could be no connection. But practice was telling us, very plainly, that there was a connection. Our job was to find out what it was. Somehow we had to reconcile theory and practice.

Perhaps there is a secondary principle here, the desire for an easy answer when there isn't one. I distinctly remember wishing that the hum would just go away, or that we'd find something quite simple, that would cure it. But it didn't go away until we did it right. The doctor who wanted my eye removed didn't have that incentive. He was in a position of authority, under government edict, that the chiropractor didn't enjoy. So he didn't have to face the true facts of that case.

We could digress into all kinds of situations, but there are enough of them right in the realm of an audio man's career. As with other branches of electronics, the audio man spends a lot of his time catching up on the latest devices: learning what they will do, and how to use them. Is that doing it right?

We suggest that it is succumbing to the weakness we just mentioned, of looking for the easy way. Who can understand what goes in those minute chips into which all the modern devices are built, anyway? All we want to know is that, if you make all the right connections to the chip, you will have a wristwatch, a computer, or an electronic organ or synthesizer.

And if something—anything—goes wrong in that minute chip, there is just no way you can fix it. You can't even see it! And at what it costs, it is easier to get a new chip and replace the whole thing. The days of pulling a defunct tube or transistor and replacing just that part are long gone. Now we replace whole chips. Or even whole circuit boards. So that is all we need to know.

There is nothing wrong with that, provided we do not forget its limitations and recognize the fact that we are taking short cuts, doing it the easy way. Then, along comes someone like John Diamond (in the January db) who comments on stress caused by listening to digitized audio. He has documented it, using methods that have been established by his profession. But audio people refuse to believe it. After all, isn't digitized audio more accurate than the older analog? That is what their discipline has led them to believe.

All the theory, made by linking up pieces of data, seems to show that Dr. Diamond must be wrong. We can simplify it into two parts. Objective tests, using oscilloscopes to examine waveforms, show that the digital processes can reproduce waveforms with far greater accuracy, and less distortion, than analog processes can; and the old Fletcher-Munson subjective tests, or updates thereof, show it to be impossible for human hearing to even detect the differences between digital and analog, if both are of the same program content.

So audio people argue that there must be some other explanation for Dr. Diamond's observations. Their theory proves there cannot be any detectable difference, so the stress he thinks he measured must come from somewhere else, or that maybe his methods of measuring stress aren't valid. There must be something wrong with the other conclusion, but they don't have to find it: they've proved their point.

I venture to suggest, with some temerity, that John Diamond may be right and that, if we persist in ignoring his findings, we may be sorry in the end. What if we'd persisted in sticking to our theory that the refrigerator clicking in and out had nothing to do with hum in our audio system? But that was a little more obvious, we must admit.

One big thing that all these theorists have overlooked is: we don't hear waveforms. What we hear is an assemblage of frequencies. Fletcher and Munson recognized that in their experiments. Now, it is true that, if a waveform is reproduced *perfectly*, then its frequency content will be unchanged. But as soon as you lack perfection in reproduction, by however minute an amount, that statement is no longer true.

Anyone who has compared analysis of a waveform by shape, with analysis by frequency content, knows that a given frequency content, even consisting of only a few frequencies, can have an almost infinite variety of shapes. So attention to precision in reproducing the exact shape of a waveform is wasted. True, this is not so in video, but we are dealing with audio, not video. Processing wave shapes in video can do wonderful things.

In video, you can generate wave shapes that produce any effect on screen you want to produce. You can superimpose pictures, make multiple images, all kinds of things. Audio is different: what matters in audio is not the precise shape, but the frequency content.

Perhaps video has a rather limited analogy with audio in the matter of color, for color of light corresponds with its frequency content. But in a very different way. On a TV screen, color is synthesized from three quite specific fluorescent colors. The colors of the rainbow, which comprise our visible spectrum, cover a whole, continuous range of frequencies. But the TV screen synthesizes colors that we recognize

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from only three basic pigments, blended in various proportions, as in color photography.

Can you imagine what audio would sound like if it contained only three frequencies? Everything would be a sort of three-note samba, with very little musical color to it. No, in audio we are in a whole different ball game. Dr. Diamond has determined that stress occurs with listening to digitized audio. Is there anything similar in video?

During the days when my eye was in that hypersensitive state, before the sight went, I was trying to find out what might be causing it. One thing I noticed was that fluorescent lighting increased the pain greatly, as compared with light from incandescent filament lamps. I was curious as to whether this extra pain, undoubtedly brought on by additional stress to my vision in some way, was caused by the color difference—the fact that simulated daylight is achieved by a selection of highly intense colors—or whether it was a function of the difference in illumination with time—the fact that fluorescent lights possess a very intense flicker, even though it is too fast for most people's eyes to be conscious of it.

By the time I had lost my sight, and then regained it, I did not have sufficient data to be conclusive. But being alerted to the fact, I have met people since whose eyes possess a similar aversion to fluorescent lighting, and have sought to collect additional data.

Returning to audio phenomena; we have established that the human ear is capable of picking up a range of sound far greater than most people's hearing has learned to interpret. Any individual can train his hearing to make discriminations between sounds that previously were indistinguishable to him. This must mean that those sounds produced different response in the mechanism of the ear all along; it was the interpretive faculty of the brain that needed training.

However, no amount of training can enable a person to hear a wave shape. He can only hear frequency content, as it changes with time. In that domain he can make almost unbelievable discriminations, far beyond those predicted by the masking data taken by Fletcher and Munson. Their data merely measured what any given individual had trained himself to be able to distinguish, and took an average.

Perhaps a misleading fact is that synthesizers generate sounds by manipulating waveshape: triangular, square and variations thereof. Maybe some are overlooking the fact that we do not hear them as those shapes. Oh, you may recognize them by associating the frequency content with that produced by various known shapes. But that does not mean that the sound you hear can be picked up on a microphone and oscilloscope, and the shape will precisely agree; almost invariably, it won't. ■

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db New Products & Services

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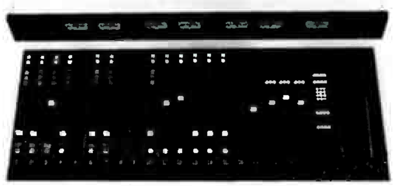


SPEAKER SYSTEM

• The Model 709 Speaker System is a wide-range, high efficiency, light-weight, portable two-way speaker system. It consists of a 15-inch woofer in a front-ported bass reflex cabinet and three piezo electric horns. The tweeters are especially designed to disperse sound at a wide angle. Power handling capacity is 150-watts of continuous program material. The 8-ohm speaker system produces about 100 dB SPL (sound pressure level) at four feet with only a one-watt input.
Mfr: Shure Brothers, Inc.
Price: $370.00
Circle 66 on Reader Service Card

Continued on page 34

What the Shadow doesn't know ...it learns.

The BTX Shadow System:

- synchronizes and controls ATRs and VTRs to sprocket-lock, subframe accuracy
- learns each recorder's dynamic characteristics to optimize machine control
- maintains minimal offset during GO TO and CHASE functions
- runs independently, allowing hands-off synchronization in all control modes
- uses standard RS 232C interface to permit present and future computer automation, including electronic editing
- available with optional control console, which displays time and status data and allows keypad entries of destinations, GO TO and FOLLOW commands, and data storage/retrieval
- priced to permit each transport to have its own Shadow

To learn what the Shadow knows, call
1 (800) 225-2253

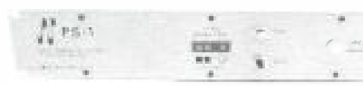
The BTX Corporation, 12 Huron Drive, Natick, MA 01760

btx SHADOW

Circle 22 on Reader Service Card

Continued from page 32

NOISE SUPPRESSION & POWER PROTECTION UNIT



• The Model PS-1 is a power line conditioning unit designed primarily for sound reinforcement systems. The transient suppressor provides protection from high voltage spikes on the power line from such sources as lightning strikes to nearby utility poles and inductive (i.e. motors & transformers) switching. The radio frequency interference (RFI) filter reduces noise from radio transmitters such as CB and from light dimmers, invariably found in night clubs, which are usually the cause of the ever-present sound system "buzz." Three neon lamps indicate relative phasing of the line, neutral and ground connections, thereby detecting improper and or grounding of the outlet in use. The latching relay prevents reapplication of AC power to loads until the power On switch is manually depressed. This feature allows the user to properly sequence power-up and thereby avoid amp/speaker damage.
Mfr: Linear & Digital Systems, Inc.
Price: $129.00
Circle 62 on Reader Service Card

DIGITAL REVERBERATION SYSTEM

• The Model 8X32 produces a wide range of natural and artificial reverberation effects suitable for studio, broadcast and live performance. The microprocessor-based front panel has separate LED read-out and control for each adjustable reverberation parameter. The 8X32 also features a bank of 32 non-volatile (they retain their contents even when the power is turned off) storage registers that allow users to store and recall 32 complete reverb set-ups, and to edit them at will. Four basic programs are available with the 8X32, ranging from a small, fast-diffusing "plate" to a large, echoing "space" simulation. Within each of these programs, 16 decay times can be selected, and the level and delay time of both the early reflective pattern and the initial reverberation may be independently controlled. The 8X32 is a rack-mountable unit measuring 3½-in. high, 19-in. wide, and 10-in. deep.
Mfr: Ursa Major, Inc.
Price: $5,995.00
Circle 63 on Reader Service Card

CONSOLES

• Two new floor standing consoles have been made available for Tascam recorder/reproducers. The CS-600 is for use with the Model 35 2B 2-track mastering recorder, which has separate housings for the transport and electronics. The CS-800 is for use with either the Model 80-8, 8-track multichannel recorder or the Model 40-4, 4-track multichannel recorder. Provisions are made to accommodate the optional dbx processors, Model DX-8, or Model DX-4. Both floor standing consoles are made of metal and feature heavy duty casters and arm rests.
Mfr: TEAC Corporation
Price: $449.00
Circle 64 on Reader Service Card

Continued on page 36

"On Line, On Budget... On Time!"

"When we decided to build The LA Studios, it wasn't on speculation. We considered what musicians, producers and engineers needed most in a recording studio. We also took a serious look at the business side of the industry. What we decided was that as a long term investment we should go first class—**MCI recorders and consoles.** We chose **Pro Audio Systems** for much the same reason. We felt they could deliver—and they did. From ground zero, **Pro Audio Systems** helped us sell our banker with necessary pro forma facts, figures, equipment specifications and total costs. They worked with our construction foreman on a daily basis. At one point two of their technicians were working around the clock to finish installation of control room A. Thanks to **Pro Audio Systems, The LA Studios** was brought on line, on budget and on time."

Jim Bredouw & Sunny BlueSkyes - Owners, The LA Studios

PRO AUDIO SYSTEMS

Exclusive MCI Representatives in the Pacific Northwest

11057 8th Avenue N.E. • Seattle, Washington 98125 • (206) 367-6800

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Continued from page 34

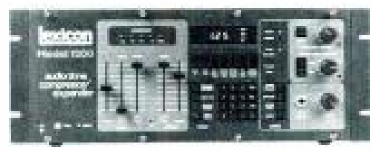
KEYBOARD MIXER/PREAMPLIFIER

• The Intersound IKP-6, a new six-input keyboard mixer/preamplifier, is the first new Intersound product to be introduced by E-V/TAPCO since Intersound was acquired by TAPCO in 1980. Each input channel is provided with two input jacks; one low gain input for high-level sources such as synthesizers and organs, and one high gain input for electric pianos and acoustic pickups. In addition, each channel features bass, treble and Auto Pad volume controls. The IKP-6 also features a master volume control, two high level outputs and two adjustable level outputs.

Mfr: E-V/TAPCO
Circle 70 on Reader Service Card

DIGITAL AUDIO PROCESSOR

• According to its manufacturers, the Model 1200 is the industry's first broadcast quality audio time compressor that can play back recorded material at faster or slower speeds without changing the original pitch. The new unit makes it possible to reduce or expand the play time of commercials or program material to meet time requirements or to accommodate tag lines and other messages. The Model 1200 is engineered for automatic use with variable speed tape recorders or for TV applications with a variable film projector/video tape recorder. Its comprehensive controls quickly achieve the precise amount of time-compression or playback/on-air time desired. Proprietary digital processing techniques allow time compression of up to 25 percent with virtually no degradation of audio quality.



Mfr: Lexicon, Inc.
Circle 71 on Reader Service Card

8 TRACK RECORDER

• Stephens Electronics, Inc. has redesigned their 811D-103A portable 8 track recorder. The improvements include an updated servo system and a frequency response of +/-1 dB from 30 Hz to 25 kHz at 30 ips. New features include a built-in 60 Hz resolver for video or film synchronizing, and the capability of operating from a 24 volt battery in the field. The portable 8 track is completely transformerless, and runs without the use of capstans and pinch rollers. It is supplied with the transport controls as a remote unit and runs at 15, 30 or 60 ips, and includes variable speeds of 10-80 ips. each channel can be assigned to automatically mute, switch to source, and record.

Mfr: Stephens Electronics, Inc.
Price: $17,580.00
Circle 72 on Reader Service Card

Continued on page 38

The New UREI Power Amplifiers

The UREI Power Amplifiers have been designed to offer the critical sound professional advanced products which extend UREI quality from our low-level signal processing all the way through to our exclusive Time-Aligned™* Studio Monitors.

After careful evaluation of other power amplifiers on the market, it became obvious that reliability has been reasonably achieved, but sound quality is often marginal. Our goal was to engineer a line of amplifiers that would offer the audio professional the reliability he demands *and* the sound quality he deserves.

The results of our quest are the following UREI Power Amplifiers:

The Model 6500 Dual Power Amplifier
Two totally independent plug-in channels, removable from the front panel, each with its own power supply and continuously variable cooling fan. Exclusive Conductor Compensation™ corrects for wire losses and delivers accurate waveforms to the speakers' terminals. 275 Watts RMS per channel into 8 ohms, 600 Watts per channel into 2 ohms.** Standard rack mount, 7″ high.

The Model 6350 Dual Channel Power Amplifier
255 Watts RMS per channel into 8 ohms.** Rack mountable, 5-1/4″ high.

The Model 6250 Dual Channel Power Amplifier
150 Watts RMS per channel into 8 ohms.** Standard 19″ rack mount, only 3-1/2″ high.

The Model 6150 Dual Channel Power Amplifier
75 Watts RMS per channel into 8 ohms.** A compact 1-3/4″ high rack mount cabinet.

Audition the UREI Power Amplifiers at your professional sound dealer and discover how good a reliable amplifier can sound.

From One Pro to Another — trust all your toughest signal processing needs to UREI.

*Time-Aligned is a Trademark of E.M. Long Assoc., Oakland, CA.

**All 8 ohm ratings are at 0.05% THD at a bandwidth greater than 20 Hz to 20 kHz.

Circle 35 on Reader Service Card

UREI From One Pro To Another
United Recording Electronics Industries
8460 San Fernando Road, Sun Valley, California 91352 (213) 767-1000 Telex: 65-1389 UREI SNVY
Worldwide: Gotham Export Corporation, New York; Canada: Gould Marketing, Montreal H4T 1E5

See your professional audio products dealer for full technical information.

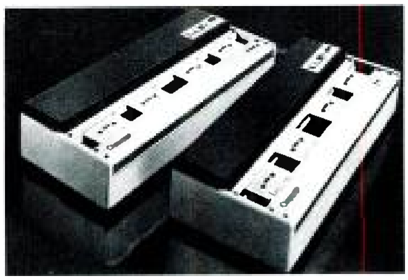
Continued from page 36

CLIP-ON MICROPHONE

• Less than an inch in length and weighing only 6.5 grams, the Beyer MCE-5 offers a full professional frequency response from 20-20,000 Hz, together with a 62 dB S/N ratio. Its non-reflective black surface passes virtually unnoticed in "on-camera" situations, and it can be powered either by a 9-18 volt battery (1.3-5ma. current) or via a 12 or 48 volt phantom supply. For outdoor applications, a detachable mesh windscreen is included. Electrical impedance is 700 ohms balanced (min. 2500-ohm input recommended), and the sensitivity is -47.5 dBm. Maximum input level is 116 dB SPL.
Mfr: Beyer
Price: $160.00
Circle 73 on Reader Service Card

CASSETTE COPIERS

• The 322 monaural and 342 stereo copiers are Pentagon's latest entries into the high speed, high production cassette copier market. Called the Producers, each has a production capacity of 75 C-60 cassettes per hour. Both sides of the cassette are copied in one pass, providing duplication of lectures, sales messages, etc. The four-channel Model C-342 copies all four channels of stereo audio simultaneously for music reproduction.
Mfr: Pentagon Industries, Inc.
Circle 74 on Reader Service Card



HORN TWEETERS

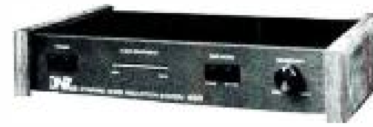
• Vortec has recently introduced its Models HF 3001 and HF 3002 precision radial and lens high-frequency transducers. Designed to reproduce the harmonics and timbre integral to clear sound reproduction at high output, both tweeters are recommended by the manufacturer for PA systems in clubs, movie theatres, and for use with musical such as electronic synthesizers and keyboards. The HF 3001 radial tweeter is employed to extend the response of mid-range 90° x 60° horns, maintain uniform pattern control at the higher octaves, and provide faster transient response and lower distortion at these frequencies. Its horn section is of die cast aluminum construction and its resonance frequency is at 2500 Hz. The HF 3002 acoustic lensed tweeter is employed to offer the same improvements for 130° x 60° acoustic lensed mid-range assemblies. Its horn is plastic with a square mounting flange, and its lens is fabricated from aluminum plate with ¼-in. spacing for smooth response to 15 kHz. Its resonance frequency is also 2500 Hz.
Mfr: Vortec
Price: HF 3001-$55.00, HF 3002-$85.00
Circle 75 on Reader Service Card

Continued on page 40

Post-Production facilities courtesy of Neiman-Tillar Associates, Los Angeles.

Continued from page 38

NOISE REDUCTION SYSTEM



• The DNR 450 is a single stage processor which requires no signal encoding. Unlike many noise reduction systems, the DNR 450 eliminates noise at the source. During playback, it constantly monitors program sound level. When the program/music level is high, it masks or covers hiss and other extraneous noise. The DNR 450 automatically determines this masking ability of the program material. As long as high frequency sound maintains sufficient amplitude to mask bckground noise, the DNR 450 extends the bandwidth (up to 30 kHz) so that all the music passes through the system. As the amplitude of the high frequency sound decreases to a point where background noise could become audible, the DNR 450 instantly reduces the bandwidth, thereby decreasing the noise level and improving overall sound quality. Total improvement in S/N ratio runs up to 14 dB.
Mfr: Advanced Audio Systems Intl., Inc.
Circle 77 on Reader Service Card

TELEPHONE REMOTE UNIT

• The Telfax TFX-31 is a miniature telephone remote unit measuring only 6¼" x 6½" x 4½", and weighing just six pounds. It is designed to feed sports, news, and interview broadcasts over standard telephone lines. The unit incorporates a complete dial telephone circuit, aural and visual ring indicators, three channel microphone pre-amp/mixer, dual level line amplifier, headphone amplifier with crown noise canceller, cassette recorder input and output channels, radio monitor input, rechargeable Ni-Cad battery pack, and AC power supply. When connected to a standard dial telephone line, or a direct loop, two-way communication is accomplished via a standard broadcast type headset. No other equipment is required. Modular and four prong phone line cords and audio patch cables are included with the unit.
Mfr: Telfax Communications
Price: $495.00
Circle 76 on Reader Service Card

SMPTE EDIT TIME CODE READER

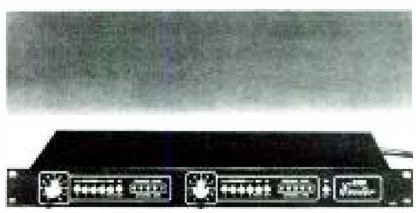
• Requiring only 1⅝ by 4¼ inches of panel space, this self-contained reader has front panel selection of either Time Code or User Bits in large .3-in. LED displays. Drop frame flag (color timing) is also decoded and displayed on the front panel. Unique circuitry is employed to "freeze" the display, useful for off-line edit decision logging, or to enable/disable the self-contained error bypass logic. The 1.75 SMPTE reader will display code from machines going forward or backward over a wide range of playing speeds. The unit is supplied wired for either 110 vac or 220 vac with external connections for alternate power sources of 6-10 vac or 8-13 vdc.
Mfr: J. S. Weiner Associates
Price: $495.00
Circle 78 on Reader Service Card

REPRODUCER AND RECORDER/REPRODUCER

• The 770 Series Reproducer and Recorder/Reproducer have been designed to meet or exceed the NAB standards for reel-to-reel tape reproducers. The 770 Series is available in a variety of configurations including one or two track, mono or stereo capabilities and tape speeds of 3¾, 7½, and 15 ips. The 770 Series is equipped with a unique professional head assembly that utilizes two torque motors, a D.C. servo capstan motor for tape drive, and is designed for simplicity and serviceability of electronic and mechanical components.
Mfr: International Tapetronics Corporation
Circle 79 on Reader Service Card

PSYCHOACOUSTIC PROCESSOR

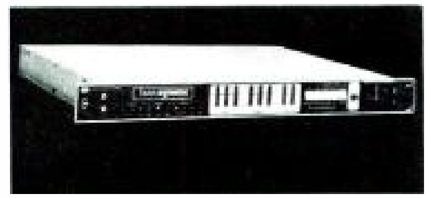
• The EX-3 utilizes techniques of time and frequency domain manipulation, preselective 180 degree phase notching and psychoacoustically-based filling of these notches to enhance the perceived loudness high-frequency content, clarity and image definition of the on-air signal. The EX-3 has 600-ohm electronically balanced inputs with r.f. suppression, and a transformerless 600-ohm balanced output capable of +20 dBm. It is phase compatible with mono or stereo applications, and the degree of processing enhancement is adjustable. Frequency response extends from DC to 50 kHz, within 0.25 dB, and S/N exceeds 90 dB.
Mfr: EXR Corporation
Price: $1,690.00
Circle 80 on Reader Service Card



MICROPHONE

• The Pro 5 is a professional unidirectional moving coil dynamic 250 ohm microphone designed for the user who has a high impedance input but prefers a low impedance model. It includes a 20-ft. cable with a low to high-Z transformer at the equipment end. The Pro 5 is particularly suited to applications that require performance dependability without a loss of sound reproduction quality. The frequency response of 50-15,000 Hz is tailored to provide full-range reproduction when used by vocal and instrumental performers at short distances. The relatively high sensitivity of the Pro 5 assures useful output and an excellent match to most mixers or amplifier inputs.
Mfr: Audio-Technica U.S., Inc.
Price: $195.00
Circle 81 on Reader Service Card

BROADCAST PROCESSING SYSTEM



• The Transdynamic Broadcast Processing System has been designed for master processing prior to AM, FM, or TV transmission. The Transdynamic facilitates separate processing of high, medium, and low frequency bands—thus avoiding modulation effects inherent in wide-band compression and limiting, thereby increasing overall audio resolution. Both 6 dB and 12 dB per octave phase compensated splitting is available with user adjustable turnover frequencies. A 30 Hz High Pass Filter can be switched in at the input to prevent severe low frequency information from entering the system. Likewise, a 15 kHz Low Pass Filter is available to prevent any unwanted high frequencies from reaching the transmitter. Internal calibration and the line-up of the full system is simplified with the built-in wide-band pink noise generator contoured to approximate the dynamics of the input signal. A stereo monitor output allows the dynamic calibration and final tweaking of the program material with the processor in the by-pass mode. The overall output of the transdynamic is metered by a fast acting PPM (Peak Program Meter) switchable between Left Channel, Right Channel, or both.
Mfr: Audio & Design Recording, Inc.
Circle 82 on Reader Service Card

AUDIO CONSOLE

• DYMA has introduced the "International" rackmount audio console for small production facilities, editing suites, news booth, or remote van. Occupying 5¼-in. of rack space, the unit features eight input modules, a headphone jack, and rotary gain pots for the master output, headphone and cue. Each channel has two inputs from the front panel and a slide fader with cue. Each preamp is switchable for mic and line level. The unit is available in both mono and stereo configurations.
Mfr: DYMA Engineering
Circle 83 on Reader Service Card

Introducing a present

Once you go through a recording session with the new ATR-124 24-channel recorder by Ampex, you'll want to go through another. Because with each new session you'll discover something new you can do. Things that you can only do with a recorder that's full of features of the future.

ATR-124 gives you the unheard of:
Time on your hands.
Which means you can use that time to give clients more of what they're paying for—your creative skills. With the ATR-124 microprocessor-based control system, you can pre-program what you want to do ahead of time so you won't waste studio time setting things up. When their time starts, you're ready to record by touching a single recall button.

ATR-124 also lets you duplicate a technique you may have used earlier in the session without having to rethink what you did. Just touch the memory button and it'll all come back to you. ATR-124 lets you rehearse what you've got in mind, without recording it, to make sure what you've got in mind is right. Tape can be manipulated faster which means you'll get the sound you want sooner. And the chance to try something "a little different." All because of the speed and accuracy that ATR-124 puts at your fingertips.

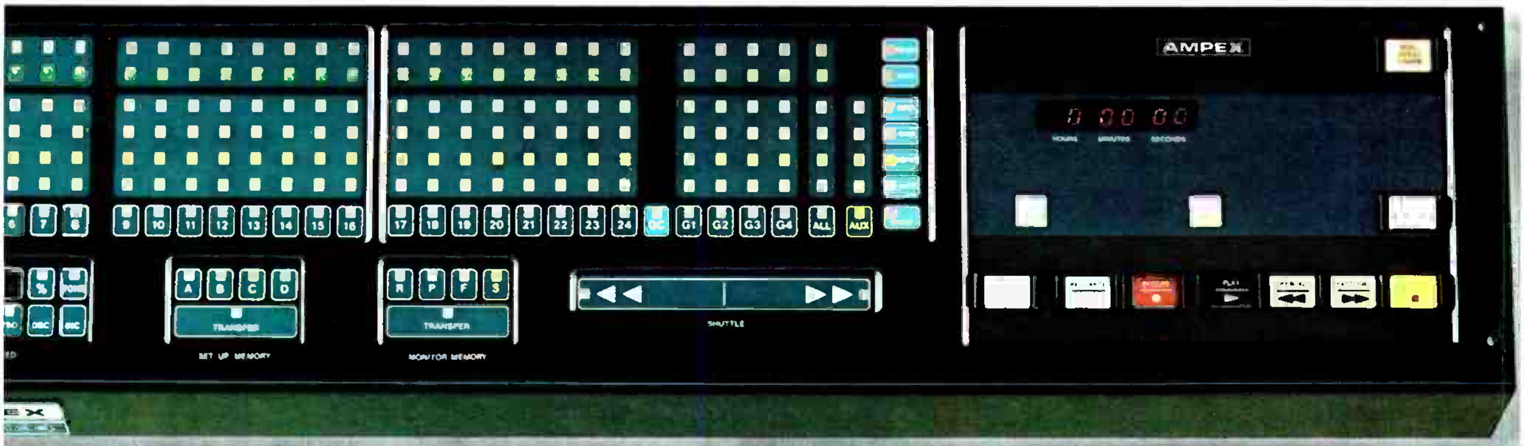
ATR-124 doesn't take away your creativity, it adds to it.
The less time spent setting up, correcting, and redoing, the more time spent creating. And when you add features that help you create to the ones that help you save time, you've got one very potent piece of audio machinery. Take the control panel for instance. It's like nothing you've ever seen. Pushpads linked to a microprocessor give you a new level of creative flexibility. Program a setup, then change it. Then change it back, all with a single fingertip.

A repeatable, variable speed oscillator for pitch correction and special effects is built in. In addition

from the future: ATR-124.

to the standard output, there is an optional auxiliary output with each channel that enhances flexibility. So don't think that ATR-124 is going to

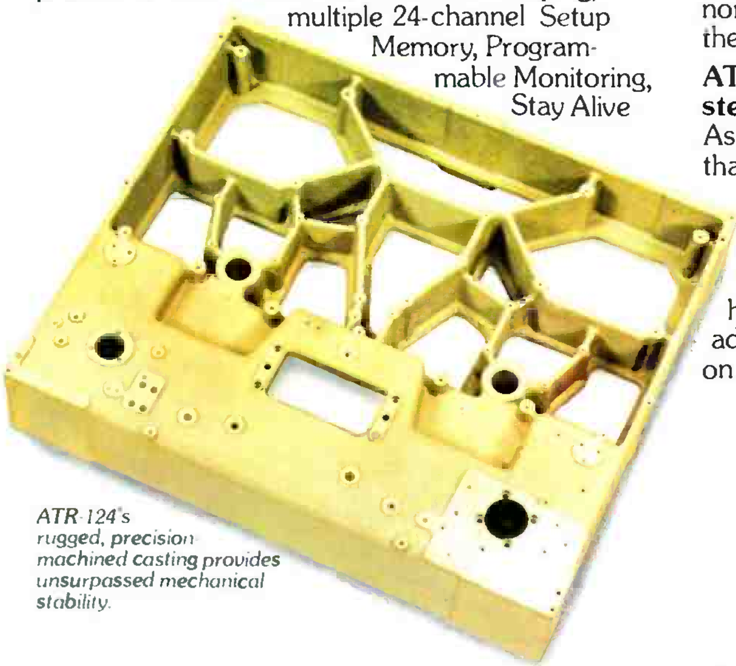
Memory, and Record Mode diagnostics. The point is this: If you like the ATR-100, you're going to love working with the ATR-124.



ATR-124's Control Panel. Speed and accuracy at your fingertips.

replace anything that you do. On the contrary, it's going to improve the skills you have, if not help you develop some new ones.

ATR-124 picks up where ATR-100 leaves off.
It's only natural that the people who brought you the ATR-100 should be the ones to bring you something better. ATR-124 offers you 24 channels instead of 4. You also get many new and exclusive features. The kind that have set Ampex apart from the crowd for the last 30 years. Features like balanced, transformerless inputs and outputs; a patented flux gate record head; 16" reel capability; input and output signal bus for setup alignment; membrane switch setup panel; fingertip-operated shuttle speed control; and microprocessor-based synthesized Varispeed -50% to +200% in .1% steps or in ¼ tone steps. ATR-124 also features microprocessor-based control of Channel Grouping, multiple 24-channel Setup Memory, Programmable Monitoring, Stay Alive



ATR-124's rugged, precision machined casting provides unsurpassed mechanical stability.

ATR-124 options.
As impressive as the ATR-124 itself.
With the addition of a built-in Multi-Point Search-To-Cue (MPSTC), you can rehearse edits and control five tape-time actuated events and be compatible with SMPTE time code. Separately controlled auxiliary output amplifiers with each channel provide simultaneous monitoring of normal and sync playback as well as all other monitoring modes. A roll-around remote control unit can also be added to the ATR-124 which contains all control features normally found on the main unit.

ATR-124's Multi-Point Search-To-Cue (MPSTC). Provides 100 cue locations.

ATR-124. Your next step is to experience it firsthand.
As you scan the points we've covered, remember that you're scanning just a small portion of ATR-124's story. We haven't even begun to discuss the accessibility of key components for easy servicing and minimal downtime, or the features we've built in to give you greatly improved tape handling. To find out more, write to us at the address shown below. We'll send you a brochure on ATR-124, our latest audio effort.

ATR-124. Pure 24-Channel Gold From Ampex.

AMPEX

Listen to the future

Audio Video Systems Division
401 Broadway, Redwood City, California 94063 415/367-2011

db Editorial

AS THE PRO' AUDIO WORLD braces itself for the coming video revolution, we can't help but notice how much press the newcomer is getting. It seems that almost every audio publication—from consumer hi-fi to pro'—is infatuated with all things video, from cameras to video systems. Hardly a month now goes by without a report on the latest in video hardware cropping up in magazines that were until recently audio-only.

Well, all of these new toys are certainly fascinating, but should they be taking up space in the pages of the audio press? And specifically, do they belong here in **db**?

The answer to that last question is definitely: Yes and No. (No hedging here at **db**!) Video will certainly have its impact on pro' audio and, if all goes well, audio will return the compliment by having its own influence on video. But to hear some "experts" tell it, the new world of video spells certain doom for all of us involved in good old picture-less audio.

We don't think so. It's been asked before: how many times do you want to watch a re-run of "I Love Lucy"? Would you pay real money to buy a videotape of "Charlie's Angels"? Well then, what would you buy, and how much would you pay for it?

First-rate movies are an obvious answer, but even here, how many times will you want to watch even your all-time favorite flick? And then compare that number to the times you've listened to your favorite audio-only record album. After you've done that, compare the prices of the two software media. And then remember that the video software is only as cheap as it is because the production costs were absorbed (hopefully) by the movie income generated by the film.

So, eventually the video disc of the movie will approach the price of the audio-only LP. Will you then buy more movies than music?

What about non-movie video productions made exclusively for the home-video market? If you think audio album production budgets are obscene now, just wait till you see the bill for a video album of the latest group on the charts.

We're certainly not suggesting that video is just another novelty act that will eventually disappear. We are suggesting that perhaps video is just another production tool that will eventually take its proper place in the world of home entertainment. Along with the personal computer and the latest in hi-fi audio, it will be a permanent part of the complete home entertainment system. However, it's not going to be the only act in town.

On the other hand, the days when the only video in town was at the local movie house, and where the audio was always dependably wretched, are over. Nowadays, both film and TV have discovered just how important audio is, and big bucks are going into the audio part of state-of-the-art A/V productions. Does that mean that tomorrow's audio chart busters will have to be great video acts as well? In the case of the super-stars, it probably wouldn't hurt, but it probably won't be the absolute necessity that some would have us believe. Instead, there will be better audio at the movies, better audio on TV, some original videodisc productions for entertainment and maybe even more for educational purposes. In addition, there should be more job opportunities in professional audio, once all these video folks discover just how important 20-to-20k really is.

On the other side of the coin, perhaps the audio recording world will gain some insights into the significance of the video experience. After all, we live in a world of surround sound *and* surround video. But as a commercial reality, reproduced surround sound is still off in the future somewhere. Surround *video*? Who knows where that is? Perhaps never, although never is a very long time.

In the meantime, it would seem that audio is not limited by as many constraints as video. Already, two-channel, 180-degree audio is on the drawing boards, and prototypes are showing up here and there. But audio engineers are discovering that, in the absence of visual (video?) cues, listeners have difficulty adjusting to 180-degree sound. (In listening tests, when dummy speakers are placed to the sides or in the rear, listeners are conditioned to expect sounds which originate around them. When these speakers are removed, the localization effects are not as pronounced. This, despite the fact that the speakers served no audio purpose. Blind listeners have no problem hearing the off-side sounds, since they are not preconditioned to reject them.)

The implications are clear. Audio and video are not unrelated media, despite what we still hear over most TV sets. We can all look forward (listen forward?) to the exciting future of sound with images—or if you like, images with sound. However, it's one more thing we shall all have to learn more about, if we are to get the most out of what's available.

This month's **db** explores some aspects of what's ahead. From automated consoles designed for video production, to getting the sound and the image together, to putting PCM audio onto videotape recorders, we offer an overview of what's in store for all of us.

As the video revolution captures more and more of the audio press, we'll try to remember that **db** is a sound engineering magazine. On the other hand, it wouldn't hurt to keep our eyes open also. ■

The Tascam 16-Track System.

Chair not included.

All this for the price of a 2-inch recorder.

If you don't need 2-inch compatibility or 24 tracks, you don't have to pay the price for expensive 2-inch hardware.

Instead, you can own the heart of a 16-track studio.

The Tascam Model 15 Mixer. 24-in 8-out 16-track monitor with a comprehensive cue system that can be fed simultaneously by 48 signals.

The Tascam 85-16 Recorder/Reproducer. A 1-inch transport with 16-track integral dbx.*

And the Tascam 35-2B Mastering Recorder. A 1/4-inch 2-track with integral dbx.

With this Tascam 16-track system, you'll get professional sound at an affordable price. And because it's all new equipment, you save even more with tax credit potentials.

Write to us today for the name of your nearest authorized Tascam dealer. He'll show you how a Tascam 16-track system can fit your studio needs and your budget.

TASCAM

TEAC Production Products

*dbx is a registered trademark of dbx inc.

Circle 36 on Reader Service Card

FRANK SERAFINE

The Sound With The Image

New technological developments will expand the horizons of sound in film. However, these advances may also bring problems for the studios further on down the road.

SOUND WITH IMAGE has progressed from its simple beginnings as a very basic musical accompaniment for a silent film, to a constantly advancing state of the art which is reaching increasingly refined levels of digital animation and synthesis. Advancements are being made at an accelerating rate, as digital research and development leads to progressively simpler operation and miniaturization. At their simplest level, microprocessors make possible computer toys and electronic games. On a more fully-realized level, they are an invaluable aid to the process of creating music and sound effects for films and video.

Frank Serafine at his own version of mission control.

Frank Serafine is a freelance sound image composer in the film, television and recording industry. Among his recent creative efforts are "Star Trek—The Motion Picture" and "The Fog."

ODYSSEY

My most recent work has been with Ron Hays' Music Image on "Odyssey," a video album which will be available in all videocassette formats by a major entertainment corporation in North America and Western Europe, as well as several different video labels in Great Britain, Germany, Austria and Australia. "Odyssey" has been released as a Videodisc in Japan by Pioneer, and has been selected as a special demonstration work by Sony Corporation of America and IBM/Discomate Associates. This 45-minute program is comprised of visual images presented with music by myself and others. The process of synchronizing the sound to the image used some of the latest developments in audio/video technology.

The visual images are created by a process called computer-assisted electronic animation, generated by waveforms, regenerated by cameras into colorizers and quantizing devices. (Some stills of these images appear with this article.) Live visual sources, such as dancers and landscapes, are electronically processed and keyed in with other effects. These elements are then edited on a convergence off-line video editing machine.

In some instances, the visuals were set to the music, and in others the music was created for the visuals. In the first case, Ron Hays would create images after listening to the composition. Later, the audio track would come back to me for sweetening and sound effects coordinated to the visual effects. In the second case, sound was produced spontaneously, layer-upon-layer, as the piece rolled.

Once the audio and video for each piece was fully realized, assembly followed. The first step was striping everything with

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SMPTE time code, in a process that saw all the audio tracks transferred from their 2-track masters to a Teac 85-16 as a rough edit, placed on-the-fly in their approximate locations, give or take a few frames. At this stage, any needed overdubs are added, as well as special sound effects such as cosmic fireworks, astral travelling and accelerations into hyperspace. This was the most creative part of the project, allowing me to draw on my sound effects work for *Star Trek—The Motion Picture*, as well as numerous shows I produced at Fiske Planetarium in Boulder, Colorado in the early seventies. There is no sound in space, of course, but the sense of motion in a soundless environment can be greatly enhanced by subtle application of the appropriate effect, using aural cues which we associate with specific situations from our exposure to analagous events in our culture through the various audio/visual media. For instance, we associate the roar of a Saturn V rocket with space travel. Viewers must often accept visual phenomena beyond their immediate scope of recognition; it becomes vital to keep in the forefront of your planning an awareness of the psychological response you are trying to evoke.

In composing musical passages, I work with a similar set of parameters. Specific tonal combinations are evocative of certain moods, a fact which has been put to great use by Indian classical musicians for centuries. And in western music, minor keys are suggetive of mournful or solemn moods; major tonalities suggest elation, happiness, awakening.

THE CREATION OF THE MUSIC

Most of the music was created with Sequential Circuits Prophet Five and Moog analog synthesizers, a modified Orchestron (a laser optical keyboard), and a Roland Digital Sequencer. The compositions were recorded at various studios around the country: some at Northstar in Boulder, some at Crimson in Santa Monica, California with the Earth, Wind and Fire horn section, and some at my own SFX Studio in Los Angeles. All the 24 and 16-track masters were mixed to a 2-track, which was used as a pre-dub in order to add additional music and effects for segues, and voice-over tracks. At this point, final SMPTE striping took place. The visual time code which is on the one-inch video master was transferred to the 16-track tape. Later on, the 16-track was transferred to an Ampex 24-track machine in order to interface with the equipment at our final post-production facility, Pacific Video in Hollywood. A BTX time code reader/generator then regenerated the original SMPTE signal on to the 24-track machine. Another Ampex 24-track was slaved to the first, using the same SMPTE regeneration process. All the tracks were subgrouped, and then transferred according to specified SMPTE edits. This gives you the ability to "slide" tracks between the two 24-track machines a specified number of frames. This was facilitated by the CMX Computers which control all the audio/video equipment at Pacific Video, with the invaluable aid of engineer Kent Gibson. Traditionally, these sort of edits have been done by the sprocket system, in which each track has its own reel of 35mm mag tape, which can be manually moved the required number of frames. The subgroups in the final transfer stage were stereo music, stereo effects, dialogue (narrative voice-over), and organic effects transferring in from an Ampex 4-track with SMPTE code on the fourth track. These organic sounds had been transferred first from a Stellavox recorder. The subgroups were then mixed through a Harrison automated 48-channel console,using a Lexicon Digital Reverb, Lexicon Prime-Time, and Eventide Harmonizers in the final mix procedure.

My current work, including that with Ron Hays, is an outgrowth of earlier efforts in the motion picture field, creating sound effects for *Star Trek—The Motion Picture, The Fog,* as well as electronic music and processing for the advertising world (commercials such as New England Bell and Zenith TV). I am finding that directors and producers are finally coming to recognize the potential of synthesis; it's a viable and cost-effective way of creating sound and music for films, television, video and multimedia productions. One synthesist can create full orchestrations by inventive use of digital sequence storage and multi-tracking.

THE EQUIPMENT

The process by which I create electronic scores and effects usually begins with the analog/digital hybrid Prophet Five synthesizer. It uses conventional analog synthesis circuitry combined with digital program storage. It has five voices, two oscillators per voice, and 40 program capability with cassette interfacing for program storage. All patching is done by internal routing; there are no external wires.

Digital/Analog converters are becoming a common way of bringing the "real" world of analog electronics into a working patnership with the computer-type capabilities of digital systems, digitizing their functions and storing the settings. Among the other D/A hybrid instruments brainstormed from the seed of the Prophet are the Korg, the Oberheim OB-XA, and the Roland JP-8. The tonal ranges of all of these are essentially equal, and while the hardware differs from one to another, all use some sort of D/A converter and cassette program storage.

At the very top of the hybrid system is the E-mu Audity, both in terms of its sonic possibilities and its price (pushing $70,000). The Audity is a computer-controlled system consisting of up to sixteen individual analog synthesizers, each on its own card, with their respective functions controllable either individually or collectively. Program storage is on floppy disk, and the system is playable from E-mu's own 16-voice keyboard or any standard one volt-per-octave controller.

Because directors occasionally have capricious ideas about what they want, the alteration of sounds in real time is often necessary. The film is put up on the screen, and the director may like the effects and mix, or he may not. If he doesn't, with a computerized synthesizer you still have the program, and you can go back to alter it accordingly. This memorization ability

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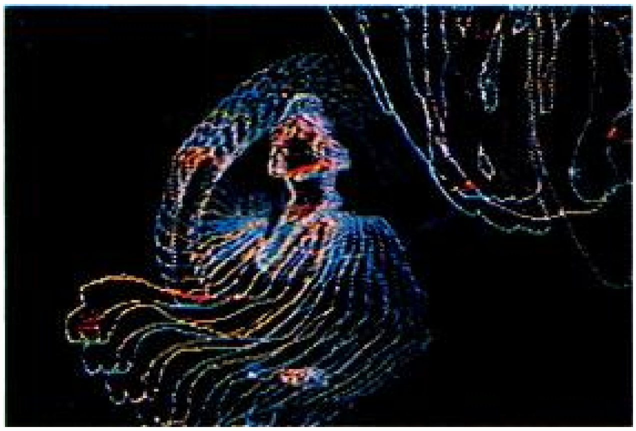

A video animated photo from a laser video disc.

can be a life-saver, when compared to having to repatch an analog instrument. I am beginning to work with the Fairlight Computer Musical Instrument. From my initial work with it, I see a vast potential for the future of synthesis in the visual media. Its most powerful feature is likely to prove to be its ability to sample a signal from a microphone or line, digitize it, and load it into waveform memory. The sound can then be manipulated in a variety of ways, including pitch change via a keyboard. This digital storing of organic sound, combined with three-dimensional waveform readout and possible SMPTE interfacing (on which I am working with Adrian Wood, general manager at the Fairlight factory in Australia), will show the Fairlight and the other digital instruments to be invaluable tools for creative spontaneity in computer-assisted sound image composing. (For more on the Fairlight CMI, see Ralph Hodges' "Zoop, Beep, Brap, Broop" in our March issue—Ed.)

The first digital synthesizer was the Coupland, developed in 1976, and other impressive instruments which followed include the Synclavier II, the Crumar-Bell (designed by Bell Labs), and the Alpha-Syntauri, ALF and Mountain Hardware systems which are designed for use with Apple Computer systems. All have interesting possibilities and bear looking into.

HOME COMPUTERS

Sound image computers are starting to explore the advancements being made in home computer programming, using computers like the Apple II and III, and the Atari. My particular interest in this area is digital electronic speech synthesis. An example of this in its beginning stages is Texas Instruments' "Speak and Spell." The most sophisticated version is the Votrax, which can store full sentence speech vocabulary in a microchip. Modules now being made by the Variable Speech Company can create pitch correction, pitch compression and frequency shifting. These devices are going to be tremendously important in developing the sounds that we'll be hearing in the next few years. These companies are also in the process of designing hardware to be controlled by the Apple, which has already been widely put to use in electronic cinema, to control cameras and projectors and to aid in computer animation. The Apple and others are going to play an important role in the interface of SMPTE-controlled video and audio, and provide convenient storage for logging tape locations.

A video animated photo from a laser video disc.

SOUND EFFECTS AND FILM

The early history of sound effects in films has traditionally involved the recording of the actual sounds at some on-location setting, or the recreation of these sounds afterwards, on a post-production Foley stage. If you were doing a Ronald Reagan shoot-'em-up, you'd go out with an old mono Nagra and chase a few galloping horses. If it was a space film, you'd be over at Lockheed recording wind tunnels. Even today, film sound has not really acknowledged the technological advancements being made. It hasn't evolved that much since the early days of 35mm magnetic recording. The same old recorders and the same old techniques are still widely in use. However, the rapid developments in the recording industry are now beginning to change this. Recording engineers and producers are crossing over into visual media, answering the new demand for higher quality sound. This rapid infusion of recording technology greatly facilitates the work of the sound image composer, who can likewise bring with him the advancements from the sound-production field when moving into sound for video and film. In addition to sounds produced with digital and analog synthesizers, the creative manipulation of organic, or "natural" sounds, fills out the resources of sound composition.

The two industry standard on-location recorders are the Nagra and the Stellavox. I use a Stellavox with a modular head structure. Its light weight can make a real difference in location situations, and its ability to accommodate different heads makes it an extremely useful, multi-functional machine. There is a basic mono head, a stereo head for stereo location work, the Stellamaster head, specially designed for high resolution musical recording (such as an orchestra), and a new 2-channel head with a middle track for synchronization or SMPTE audio/video interlock tones.

Signal processing is a key element in the creation of sound effects. On the video album, we needed a thunderclap to match flares of light, and we created it by using the Moog and the sound of a real thunderclap. In tracking the sound, we considered the physics of light and sound and the time discrepancies that one encounters when experiencing lightning and its ensuing thunder. There was a sensation of considerable distance from the source of the light. However, had the sound been delayed to suggest that distance, the psychological impact at the moment of flare would have been lost. We chose to have only milliseconds of delay at the onset of the effect, but to draw the sound out by a variety of delay methods to suggest the impression of spatial vastness. First, we used a basic SMPTE trick: advancing each overdub five to ten frames for the time units desired. Then, a DeltaLab Acousticomputer was used to get a deep phasing effect suggestive of the phase distortion one might experience with such a sound. Finally, the Lexicon Digital Reverb was used, in its chamber reverb preset mode. This was also used on the musical sections, particularly the acoustic orchestrations, which had been recorded dry.

It's simple enough to provide a single sound in a single form, but this commits you to that particular sound prior to

The Fairlight CMI.

The Oberheim OB-Xa.

dubbing. I will often provide a sound in several different versions, each with a different set of treatments. I like to have this latitude to mix numerous options in a variety of ways toward the unknown end.

My signal processing equipment currently consists of the DeltaLab Acousticomputer, a Maxon AD-230 Analog Delay, several Scamp modules, Roland's Dimension D, Vocoder and Pitch-to-Voltage Synthesizer, and the EXR Exciter III.

Dialogue sound processing has become a common practice in many films. I recently did some work for Francis Ford Coppola's Zoetrope Studios, processing dialogue and creating effects for one of their innovative electronic storyboards (essentially, a pre-shoot video rehearsal demo). I ran all dialogue through the Exciter, which is a psychoacoustic enhancing instrument that creates a sensation of greater dimension in the highs, adding clarity and transparency without any gain in decibels.

MIXING

Mixing for motion pictures is generally a three-man job, divided into music, sound effects and dialogue. There are also three basic mixing modes: stereo, 6-channel (for "sensurround"), and mono for TV and video release. In contrast for mixing for records and television, this type of mixing has traditionally been done on sound stages the size of a full length theatre, in order to custom-fit the sound for actual theatre application. Mixing for video can either be full-bandwidth stereo, as in the case of "Odyssey," or similar to television, in which you have to consider the crusty old speaker in the TV set. When doing this type of mix, I monitor on Auratone Cubes and the speaker in a Panasonic video monitor, and A-B between them, adding the EXR Exciter for clarity.

Mixing for theatrical film use can be equally problematic. The physical format of film was designed to serve the picture, with little attention paid to accurate sound production. The Academy Curve, which is used in the re-recording stage and limits the frequency spectrum, and the erratic nature of the transport mechanism in the projector, need to be considered in advance and compensated for wherever possible. I mix on small speakers so that when the mix is played back against the picture, it retains something of its original effect. Getting it to sound great on high-quality studio monitors can be pointless, because it'll never be heard that good again. It's not the people in the field, but rather the fact that the tracks get so manipulated between production and the showing of the film in Bakersfield or Poughkeepsie on an old projector after innumerable transfers and in optical form.

SMPTE

A brief overview of SMPTE is in order. (SMPTE stands for the Society of Motion Picture and Television Engineers.) SMPTE time code synchronizing techniques had their beginning in videotape technology and the need to synch up video machines. The code itself consists of a stream of unique pulses, thirty to the second, called frames, and identified in groupings of hours: minutes: seconds: frames. Each individual frame is specified by its own time address. A master SMPTE generator/reader, such as the BTX model 5400, can read the code from an existing master tape and regenerate the identical code on to the "slave" tape that is being synched. This was put to extensive use on the video album.

SMPTE is a flexible solution with many possibilities. It gives you the ability to link two multi-track machines together, providing unlimited tracking capability. In extreme over-dubbing situations, mixing a composite pre-dub on to the slaving multi-track frees the original from wear and gives you an instant reference mix. When you are ready for final mixing the machines are interlocked. The freedom provided by this pyramiding technique has already proven to be tremendously useful for me. On my Teac 85-16 I track on the first fourteen channels, laying a 60-cycle tone on fifteen for resolving and to act as a buffer in case of problems with the SMPTE track. The SMPTE track itself goes on channel sixteen, though this can lead to problems if the edge of the tape is damaged.

In most cases, considerable modification will be necessary for SMPTE compatability. My VTR is a Sony VO-2600 with a conversion kit consisting of a playback head and a circuit board. This head is mounted on the chassis next to the head block and is always in contact with the tape, thus providing a constant signal. This allows the tape to be read in shuttle modes (fast forward and rewind), so that the slave machine can chase the master machine accordingly. This modification was made so that I could interface my 16-track with a BTX 4600 controller and 4500 synchronizer.

Wider use of synthesis and SMPTE techniques, along with the other new technological developments, may create a dilemma in the often political world of studio production. Objections are sure to be raised by unions and middlemen. The impact is already being felt quite strongly, with entertainment corporation executives speculating in print on job losses due to the digital revolution and simplification of operation. The traditional trade categories are no longer necessarily valid, as innovators begin to cover more aspects of a job with better results, lower costs, and streamlined methods. The passing of the old methods will not be easy for many in the business. ■

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BARRY J. ROCHE

NECAM II

A detailed look into the workings of the NECAM System.

THE NEVE COMPUTER ASSISTED mixdown system—NECAM—was first introduced in 1976 and was an immediate success. Neve had provided what engineers and producers wanted, a sophisticated yet simple to operate automation system which would also relieve them of many routine tasks. Essentially, NECAM provides the mixing engineer and producer with more hands and a better memory.

Although NECAM was designed principally for the music recording industry, we found an increasing number of requests for NECAM systems from television and film studio post-production facilities. We recognized that although the NECAM System satisfied many of the demands of post-production, a number of enhancements might permit an even broader coverage of these requirements. Accordingly, in 1979 an enhanced system was introduced and given the name of NECAM II. Still in widespread use, the NECAM—or NECAM I as it has been dubbed—will be described first, followed by the additional features of NECAM II. All NECAM I features, however, are incorporated in NECAM II.

Barry Roche is executive vice-president of Rupert Neve Incorporated, Bethel, Connecticut.

NECAM I

NECAM, which can be fitted to any Neve Console, uses a powerful mini-computer and dual floppy disc data storage to memorize and reproduce the console fader positions and associated mutes for up to 999 mix attempts. In addition, the NECAM System has total control of the multitrack tape machine which is reproducing the tracks for the mix attempt. SMPTE time code is reproduced from one track of the multitrack tape (usually the last track) and is used as a time reference for the entire system. Inasmuch as fader positions and mute data are stored on floppy discs and not on the multitrack tape, no slippage of mix information can occur on successive mix attempts. A console fitted with the NECAM System looks very similar to a manual console, except for the NECAM control box, which is actually a computer terminal in disguise (see FIGURE 1). However, the familiar letter keys of the terminal have been replaced by keys with even-more-familiar names like play, wind, keep, group, etc. The control box's alphanumeric display communicates and instructs the operator via a series of messages in plain English. The NECAM software package eliminates the need to be a computer programmer or keyboard operator to run the system.

MIXING

Mixing on a NECAM console is much the same as mixing on any ordinary console, with the exception that all fader movements and muting can be memorized. On replay, the faders actually move under the control of built-in servo motors and the mutes switch on and off, exactly as they did during the

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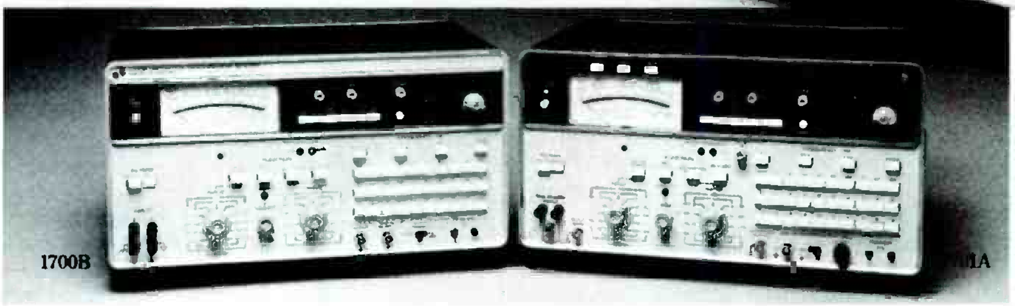

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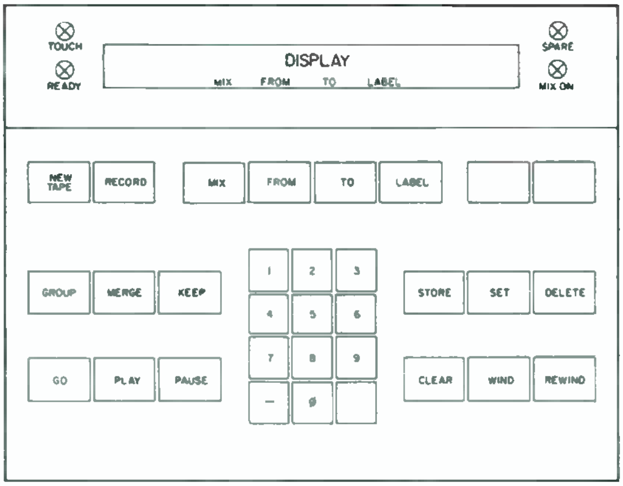


Figure 1. The Necam I control box.

mix. This is a little eerie at first—but when engineers see their own mixdown being reproduced exactly, they realize that they are working with a very sophisticated and powerful mixing aid.

The basic NECAM philosophy is that all functions should be instinctive. In other words, they must be performed just as they would on a non-NECAM console. For example, to update a fader position, simply move the touch-sensitive fader to some new position. The mere touching of the fader puts the system into the update mode. This movement is memorized by the computer and stored for later replay.

As part of the NECAM system, a sophisticated tape auto locator uses NECAM "LABELS." These are simply numbers from 1 to 999 that can be established at any point during a mix by pressing the LABEL key on the control box. As the key is depressed, the computer allocates the next available label and memorizes the time code reproduced from the tape machine. LABEL points can be used as starting or stopping points for the tape machine and also as editing points. The LABEL numbers are substitutions for the eight-digit SMPTE time code which gives hours, minutes, seconds and frames.

Because it is possible to memorize up to 999 mixdowns, NECAM can use a sophisticated MERGE function to provide a final mix from any combination of mixdowns by "BUTT SPLICING," "SELECTIVE TRACK MERGING," or by a combination of both. To carry out a BUTT SPLICE, all mix data located between two label points of one mixdown is combined with mix data from another mixdown located between another pair of label points. The process continues

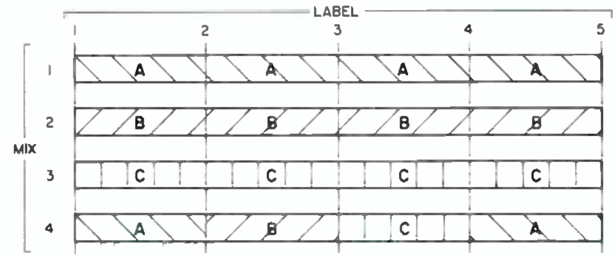


Figure 2. Mix 4 is a "butt-splice" merge of mixes 1, 2 and 3.

until an entire mix is compiled (see FIGURE 2). Similarly, SELECTIVE TRACK MERGING is carried out by combining individual tracks from different mixdowns, until all desired tracks are selected (see FIGURE 3). At the end of a merging sequence, the control box KEEP key is pressed. The computer then automatically allocates the next mix number to the "merged" sequence and stores it as a new mix on the floppy disc. During the entire merging operation the multitrack tape remains stationary, as only data is being manipulated within the computer system.

Even after a complicated merging sequence involving data from many mixdowns, the data from those mixes remains memorized on a floppy disc called the "DATA" disc. Information will only be removed from the DATA disc if a mix is deleted. To delete a mix, the mix number is entered on the control box, followed by the DELETE key. Deleting a mix will provide more storage space on the data disc. Not only is it possible to copy the entire data disc using a copy program, it is also possible to copy selected mixes onto a new data disc with the selected mixes as the starting points for subsequent mixes.

GROUP FUNCTION

A "GROUP" function is also available on NECAM I. This facility allows any combination of faders to be electrically linked, and, when any one fader in the group is moved, the remaining faders in the group also move, thus retaining their relative balance. As mentioned earlier, the alphanumeric display in the control box communicates and instructs. For example, when the GROUP key is pressed a message will appear on the alphanumeric display saying PLEASE TOUCH REQUIRED FADERS. The faders required to form the group are then touched, followed by the GROUP key. From that point on, until the group is cancelled, all faders touched will remain grouped. It is possible to alter the balance within the group by touching two grouped faders at the same time and altering the balance.

Figure 3. Mix 3 is the "selective-track" merge of mixes 1 and 2.

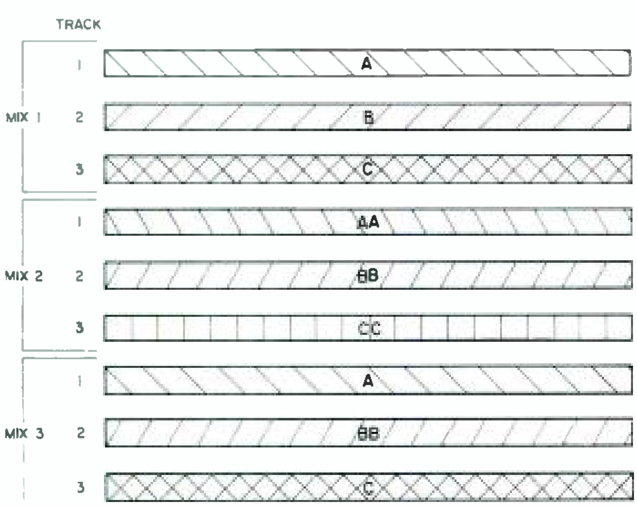
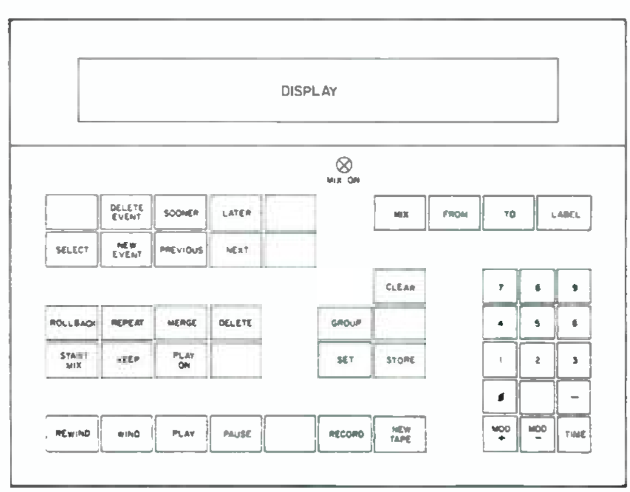


Figure 4. The Necam II control box.



Automated fader mutes are provided with each NECAM fader via a three-position switch which allows the fader to remain ON, OFF, or in the AUTOMATIC mode. During a mixdown, all mute switch positions will be recorded and replayed if the mute switch is placed in the AUTO position. The only other switch associated with each fader is the NORMAL, RELATIVE and MANUAL switch. In the NORMAL position, the fader will reproduce all movements made during the mix attempt. The RELATIVE mode enables a level offset to be introduced to all the memorized fader movements and the MANUAL mode allows the operator to override the information being fed back from the computer during a mix replay, even if the fader is not being touched.

THE CONTROL BOX

The first significant difference between NECAM I and NECAM II is NECAM II's more elaborate control box (see FIGURE 4). The control box contains a number of additional keys and a four-line alphanumeric display, all of which emphasize the differences between the music recording application and the post-production application. FIGURE 5 shows a typical multitrack mixdown/remix configuration for NECAM II. NECAM II in this case is controlling a multitrack tape machine whose record functions are under NECAM control. This is not unlike a typical NECAM I installation which has an auto record facility fitted. The NECAM I's auto record facility switches the multitrack tape machine into and out of record at programmed times. These programmed times are generally determined "on the fly" and are, in effect, consistently accurate punch-ins and punch-outs always occurring at the same time code points. The more typical post-production dubbing suite with facilities for multitrack mixdown and effects cueing is shown in FIGURE 6. In this case, the NECAM II System is controlling a video tape recorder (VTR) with various audio tape machines (ATR) slaved to it via a synchronizer. The NECAM II requires a different software program to control the VTR which in many cases, is a ¾-in. video cassette recorder and requires a different sequence of commands than that of an ATR when going, say, from play to an auto locate function.

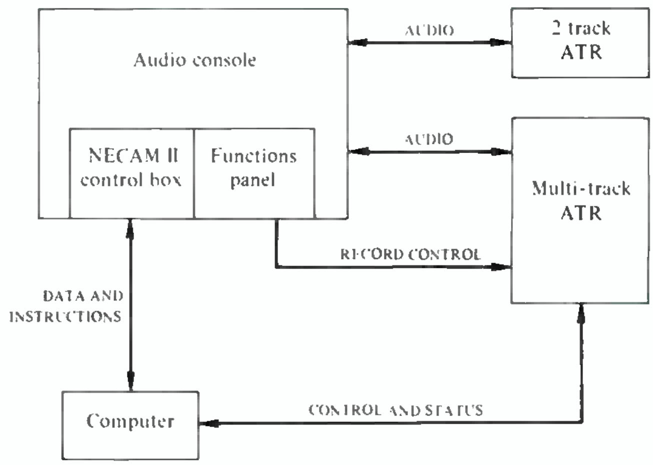


Figure 5. The flow diagram for a typical TV systems multitrack mixdown/remix.

As shown in FIGURE 6, NECAM II is now becoming a bit more complex. It is now possible to dub to and from the VTR, the ATRs and various effects machines. The objective now is not only to produce a composite audio signal but to produce an audio signal that will be mechanically and artistically correct when combined with a video picture. The video from the video tape machine is generally the final edited version of the program to be dubbed, and in some cases the multitrack ATR could have the original soundtracks from the video tapes before editing. (Unfortunately, this doesn't often happen and therefore restricts the audio creativity at the video edit points. More often the audio on the edited video tape is the work of a video editor.) It is this soundtrack then that requires rerecording and mixing with the original soundtrack, sound effects, music, etc. NECAM II is ideally suited for this application as the composite soundtrack can be compiled with the great precision which is required when being synchronized to a video picture.

OTHER FEATURES

NECAM II offers a number of features that can ensure this precision. In addition to the contol box (FIGURE 4), another panel provided with NECAM II is called a "function panel." This consists of up to 64 momentary switches which control a relay changeover—on or off—as indicated by an associated LED. The switching on or off of these relays is called an "event." There are up to 999 events that can occur with an accuracy within one frame (1/25th or 1/30th of a second) relative to the VTR picture.

In music recording, reference points are established by means of LABELS which are equivalent to the tape's time code, and generally, it is not necessary to know what that time code is. (It is, however, possible to call up a LABEL's time code on the display.) In post-production applications, the SMPTE time code is more routinely used as a means of communication between man and machine. In addition to being able to generate labels "on the fly" as in NECAM I, NECAM II permits labels to be established at specific time codes. This is achieved by

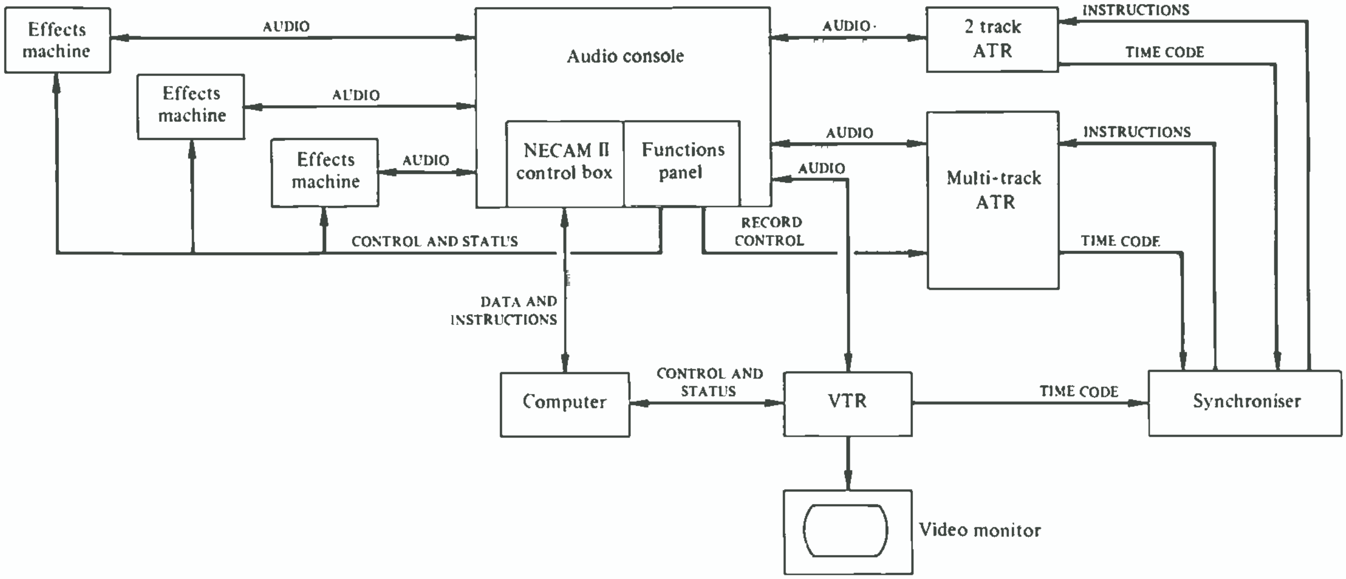


Figure 6. The flow diagram for a dubbing system, with facilities for multitrack mixdown and special effects cueing.

entering the time code on the display and then pressing the LABEL key.

The time code from the video tape machine that is used to control the NECAM System and the synchronizer is generally superimposed at the bottom of the video monitor. Either this time code or a time code from the video edit list can be used to program label points or events.

A function is selected by entering the number of the function on the control box keypad and then the SELECT key. A time code is then entered via the keypad and depressing NEW EVENT will result in the creation of an event for the selected function at the time code specified. The display will then show the event created together with its switch state (on or off). An event can be entered "on the fly" while the video tape machine and system is in PLAY under NECAM control. Operation of a function switch will cause an event to be created for that function at the time code point on the tape currently passing the tape heads. During the replay of the mix, the status of the function circuits is always displayed by the LEDs associated with the function switches. The facility is also available for control box display of all the events and their time codes, as they occur. If required, just the events for a particualar function can be displayed, or none at all. Once a function has been selected, the keys PREVIOUS, NEXT, SOONER, LATER, MOD PLUS and MOD MINUS, become available. PREVIOUS and NEXT result in a display of the previous or next event for the function selected, relative to that shown on the display. If a function is selected and then a time code entered, pressing PREVIOUS (or NEXT) will result in the display of the event of that function immediately prior to (or after) the time code entered. When a function has been selected and an event is shown on the display, pressing SOONER (or LATER) will result in the event being moved backwards (or forwards) in time by one frame at a time. The keys marked MOD PLUS and MOD MINUS may be used to move an event by a time that is larger than would be practicable by repeated use of the keys SOONER or LATER. When the event for a selected function is displayed and a time code entered as an offset, pressing MOD PLUS will result in the offset being added to the time code of the event shown and MOD MINUS causes it to be subtracted. It is clear, then, that effects cueing can be very precise under NECAM II control.

Needless to say, the precise timing of effects and cues requires repeated previews of the video and audio material. This is easily achieved using the ROLLBACK facility, which causes the tape to be rewound a preset distance; the mix is then resumed by pressing the PLAY ON key. The present distance may be altered by a time code offset directly before pressing ROLLBACK. If, however, an offset is entered which would cause the tape to rollback to a point beyond the start of a mix, the tape will rewind only to the start of the mix and resume play from there. When used after ROLLBACK, pressing the REPEAT key will cause the tape to rewind to the point to which the tape was last rolled back and the mix repeated from there. The REPEAT key can be pressed as often as necessary to perfect events cueing or balance.

Sometimes during a dub or a mix, it may be necessary to search the tape while under NECAM control. This is possible with NECAM II by using the WIND and REWIND keys. When either of these keys are pressed, a SUSPEND message is displayed which, in effect, means automation facilities have been suspended while the tape is being repositioned. This facility is called ASSIST REWIND. Only the PLAY ON or CLEAR keys will cancel the suspension and cause an UPDATE message to be displayed, while NECAM II updates the mix to the point in time when the PLAY ON key was pressed. When the mix update is completed, the tape will continue to play.

It has now been shown that NECAM II can exercise far more control over the tape machine it is working with than is possible with NECAM I. This machine could either be an audio tape recorder, as shown in FIGURE 5 or a video tape recorder, as shown in FIGURE 6. Although retaining all the NECAM I features such as grouping and merging, NECAM II provides the moveable events via the function switches, rollback and assist rewind, which are essential in post-production applications.

So far this year—Motown in Los Angeles has upgraded one of their three NECAM I Systems to a NECAM II System, and Skagg Broadcasting has taken delivery of the first 8108 console to be fitted with NECAM II. The BBC Sypher II, ITN, ATV, and The Music Center in the United Kingdom, ABC in Sydney, Australia, and Bavaria Atelier Film Productions in West Germany—are now seasoned NECAM II users and report great success with their advanced facilities. ■

CURTIS CHAN

The Sony Digital Picture

A look at Audio Recording and Editing on Videotape.

THIS TECHNICAL PAPER presents an overview of the technology behind the Sony PCM-1600/PCM-1610 digital audio processor. Subjects covered include the encode/decode sequence of the processor and the determination of the sampling rate. The principle advantage of digital recording over analog are: no generation loss, increased dynamic range, low harmonic and intermodulation distortion, freedom from wow and flutter, no print-through or tape noise, and perfect phase relationships.

The principle advantages of digital recording over analog recording are: no generation loss, increased dynamic range, low harmonic and intermodulation distortion, freedom from wow and flutter, no print-through or tape noise, and perfect phase relationships.

THE ENCODING PROCESS

Referring to FIGURES 1 and 2, an analog signal from the output of the mixing console is fed into the audio section, utilizing Deane Jensen op-amps. The signal is then passed through a low-pass filter (LPF) and buffer stages. The LPF is inserted before the A/D chain to prevent aliasing components from being generated in the processor. Because the filter is of a 13-pole design (giving a 130 dB/decade cut-off at fc), band restriction is sufficient to prevent aliasing noise from being generated.

Sampling serves to convert the analog signal into pulse amplitude modulation (PAM) signals at a fixed time interval. The sample interval is the reciprocal of the sampling frequency and is fixed for a given system.

The sampled signal is then sent through the 16-bit linear A/D converter where the signal is quantized and encoded. Quantization refers to the process where each sampled analog level is represented by an assigned value, which is substituted for the analog signal.

When linear quantization is used, the number of sampling levels is given by 2^n, where n is the number of bits used in the system. The dynamic range in dB is given by 6N+ 1.76. For a 16-bit system, the number of sampled increments are 65,536, steps and the theoretical dynamic range is 97.76 dB. For simplicity, we will use a 3-bit system as shown in FIGURE 2, with 2^3 or 8 levels of quantization. The analog signal, which has now been converted into discrete values by sampling and quantizing, is further converted into pulse codes, which represent the amplitude of each sampled interval, i.e., PCM (Pulse Code Modulation).

The coded signal is then sent through the encoding.interleave circuitry. It is within this process that interleaving, a method to disperse drop-out errors by changing the sequence of information words in the recording process, is used. In conjunction with interleaving, an error correcting code called CWC (Cross Word Code) is also used to encode the digitized signal. Using CRCC (Cyclic Redundancy Check Code), the probability of detecting and decoding erroneous data is $1 - 2^{-n}$, or 99.9985%.

Next, the encoded signal is translated into a pseudo-video signal compatible with NTSC standards. This pseudo-video signal is recorded onto a videotape recorder (VTR). The reasons for using a VTR as the storage medium for encoded signal are two-fold. Firstly, the transmission rate of the PCM-1600/PCM-1610 is 3.5795 M bits/sec., making the VTR the logical choice because of its ability to store information in the 4 MHz range. Secondly, VTR standards have already been set, making the software compatible with other companies using the same format. FIGURE 3 shows the encoded video signal using the interleave and crossword code error correcting schemes.

THE DECODING PROCESS

Referring again to FIGURES 1 and 2, in the decoding process, the stored video information is played back through the VTR where the composite video signal is decoded. The sync and data separators separate the sync and data components and the data words are sent through the de-interleave crossword decoding circuitry, where error correction takes place. Although an in-depth analysis of CWC and interleaving are

Mr. Chan is the Western regional engineering manager, Sony Digital Audio Division.

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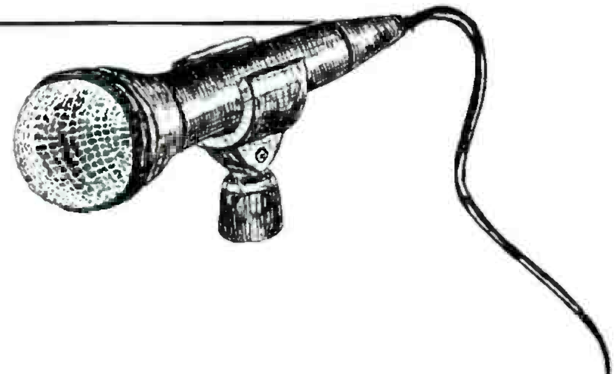

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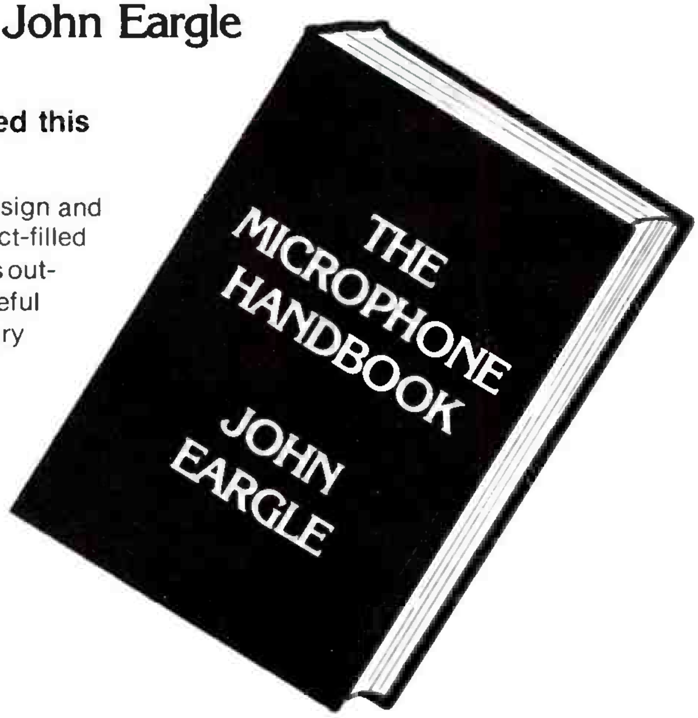

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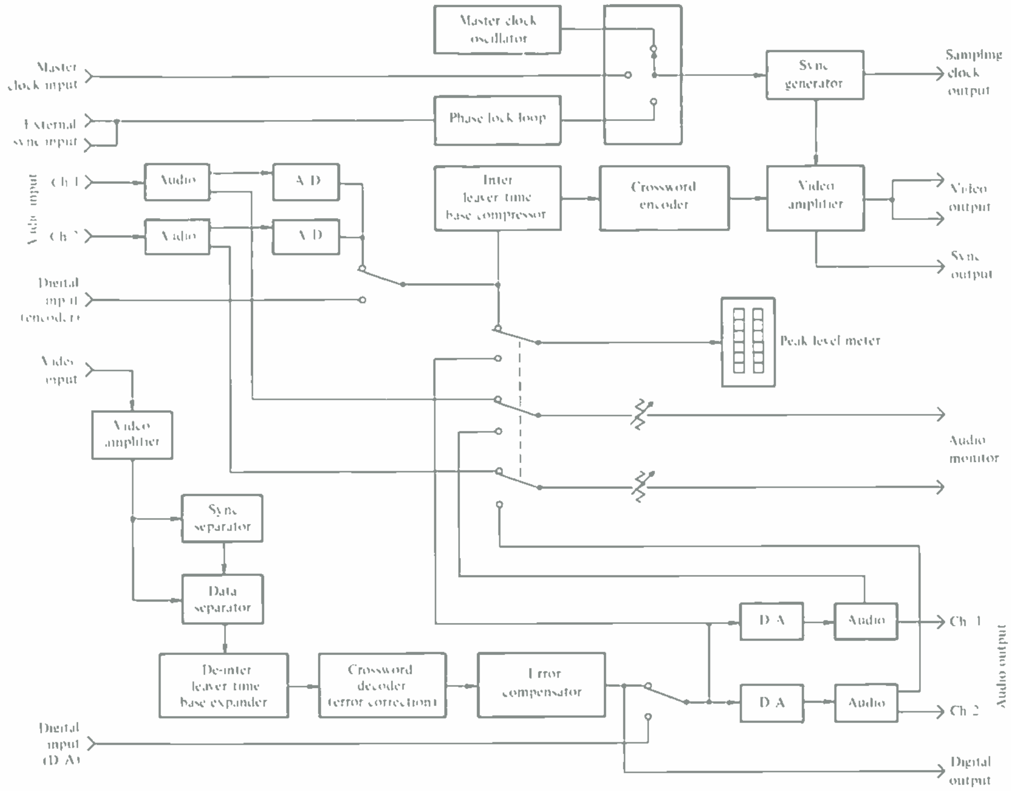
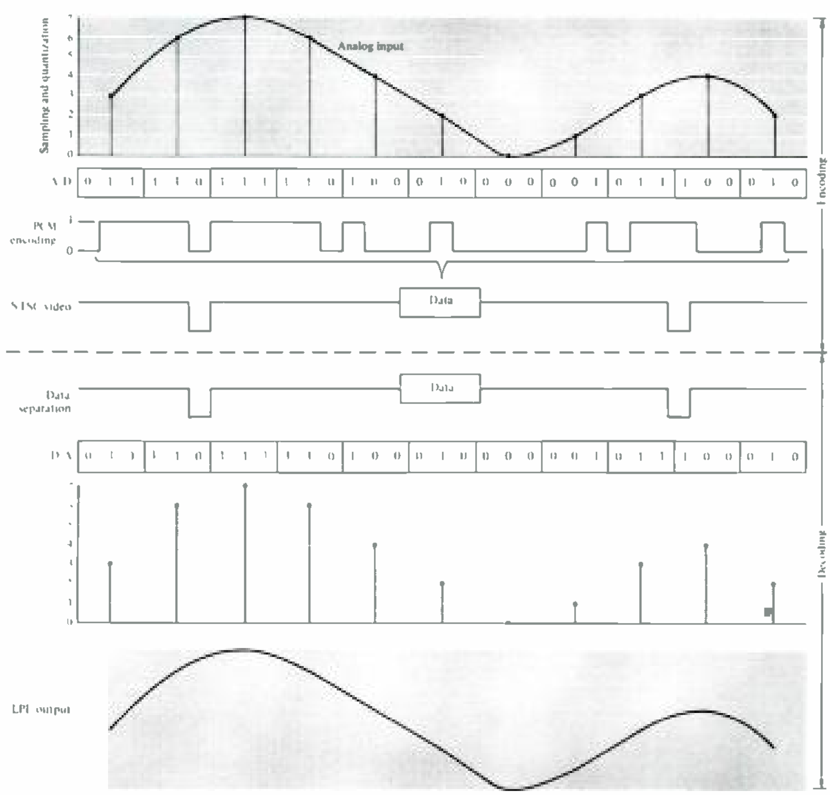


Figure 1. Block diagram of the digital audio process.

Figure 2. The analog-to-digital-to-analog process.



CBS staff producer Bruce Botnick editing Kenny Loggins' digital tapes at Digital Magnetics, Los Angeles.

not the main objectives of this article, the distinguishing features of the code are listed below:

1. The code is effective both for burst and random errors. If errors exceed the correctability of the code, minimum numbers of words are renounced so that interpolation can take place.
2. Because CWC discriminates and corrects errors with words (16 bits) instead of unit bits, large block code implementation of hardware can be easily achieved.
3. By virtue of interleaving, this code can completely correct burst errors within 2,240 bits (11.7 H Lines) in one interleave block, and burst errors within 4,480 bits (23.3 H Lines) can be compensated by one-word interpolation.

The corrected data signal is then clocked into the D/A converter stage where the digital signal is converted back into analog form. From here, the analog signal is sent through an LPF to smooth out any minute irregularities due to the reconstruction of the waveform. The final output, once again utilizing Deane Jensen op-amps, is stepped up in gain with an adjustable 8 dB variance.

As a final note, since the PCM processor has digital input/output features, direct digital-to-digital transfer of program material can be easily accomplished with the use of the DAE-1100 digital audio editor. The advantage is that the program material suffers no generation loss, because the data is in digital form. In addition, the digital signals can be transformed to another format as when the PCM-1600/PCM-1610 (16-bit) is used to dub to the PCM-100 (14-bit). The signals can also be encoded with new effects, as with the use of a digital reverberator (the DRE-2000).

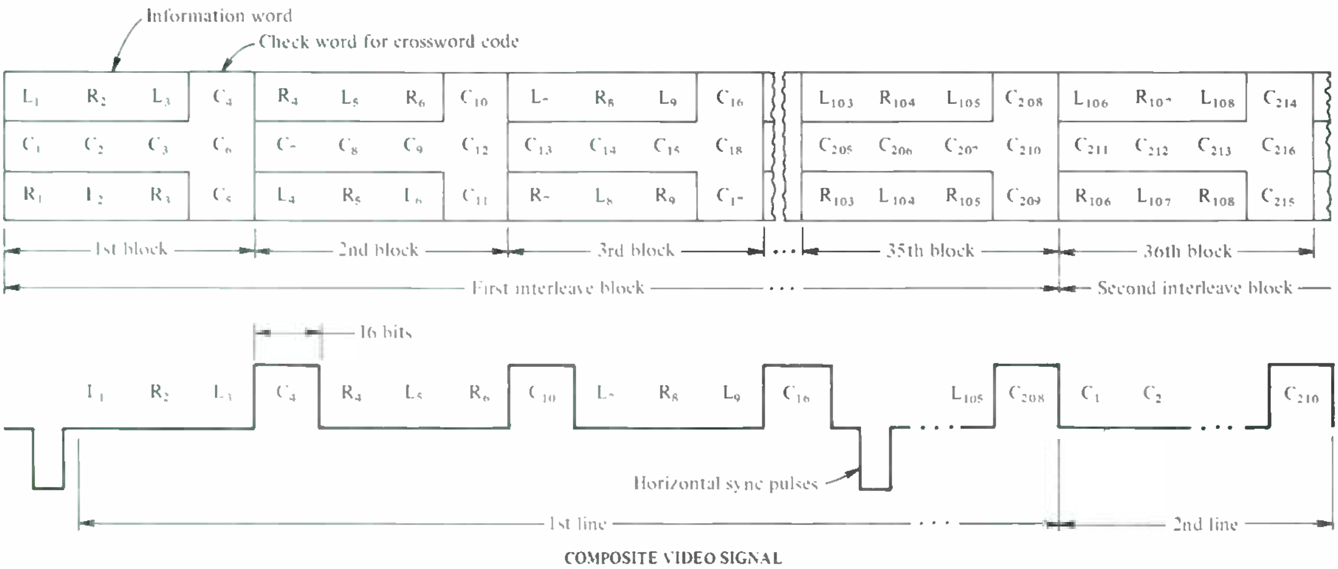
DETERMINATION OF THE SAMPLING RATE

The following section describes the conditions under which the sampling rate was chosen. If we assume a 20 kHz bandwidth for recording and playback, and the practical problems in making filters to prevent aliasing noise, the mimimum sampling frequency (f_S) is 42 kHz.

Without modification to the VTR and in keeping with video standards, no PCM data can be inserted during horizontal and vertical blanking intervals. Also, head switching time of helical VTRs take a maximum of 6½ H Lines and 2 H Lines are used for system control. This restricts the number of H Lines (H_{PCM}) per field of PCM data which can be inserted to no more than 248.

In terms of PCM data, one word is one sample, which includes the data from two audio channels, and the number of bits for one word is 26 to 32. Additional bits are used to compensate for drop-out errors.

Considering the frequency response and jitter characteristics of VTRs, it is difficult to insert more than three words in a 1 H Line. Non-integral numbers of words, though feasible, are not recommended because of the complexity of the hardware design.



1 interleave block = 35 blocks
= 105 information words x 2 channels
7 interleave blocks = 245 blocks
= 735 information words x 2 channels
= 1 video field

1 word error = 100.0%
2 word error = 100.0%
3 word error = 46.8%
4 word error = 15.8%
5 word error = 7.6%

Block size = 12 words
Information words = 6 words
Check words = 6 words
Burst error correctability = 4 words
Random error correctability = Full decode

Figure 3. The PCM-1600's interleave system and the composite video signal.

In view of the above data, the number of words (W_{PCM}) that can be put in one field of video is

$W_{PCM} = 3 \times H_{PCM} = 744.$

The resultant maximum sampling frequency (f_S) is

$f_S = W_{PCM} \times f_V = 44.5954$ kHz,

where

$f_V = 59.94006$ (NTSC vertical sync rate).

The master clock frequency should be an integer multiple of both the sampling frequency and the TV equalizing pulses, $f_{EQ} = 31.468528$ kHz. In addition, the master clock frequency (f_M), must be N times the least-common multiples of f_S and f_{EQ}. N is the number of bits for one word (N = 32). Therefore $fm = N \cdot n \cdot f_S = N \cdot m + f_{EQ}$. Sampling frequencies which meet these conditions are indicated in FIGURE 4. The table also shows the ratio of f_S to f_{EQ}, where m and n are integers.

In keeping with the constraints shown previously, sampling frequency No. 4 is an ideal candidate. For the PCM-1600/PCM-1610, f_M is four times the sub-carrier rate, or 4 × 3.5795 MHz = 14.31818 MHz. The bit clock rate is 1.1014 MHz, and the word clock is $f_M/325 = 44.056$ kHz.

Sony DAE-1100 Digital Audio Editor.

THE EDITING OF DIGITAL AUDIO TAPES

In a conventional analog system, editing is performed by manual splicing of the magnetic tapes. In digital audio, editing is accomplished by the recording of electronically selected data in digital-to-digital dubbing. When compared to the analog method, the ideal digital audio editor should have equal or better functions. For example:

1. The edit point should be easy to locate.
2. Editing accuracy should be better than 1/100 sec.
3. Amplitude changes should be variable as the producer desires.
4. Equal functions should be available in both the rehearsal mode and in actual editing.

A significant advantage of digital editing is that the original tape can be used with no fear of erasing or damaging the tape.

FIGURE 5 shows how digital editing is performed. First, decide the fade-in points on the master tape. Then, digitally dub from the master tape to the editing tape.

As shown in FIGURE 6, the editing system consists of two or more VTRs (one operates as a recorder, and the other(s) as playback), one digital audio processor, and one digital audio editor. Two playback VTRs permit the automatic location of the edit point on one player while editing is performed from the other player. Editing functions are the same in both players.

LOCATION OF THE EDIT POINT

The editor incorporates a memory circuit to store digital signals. This allows easy edit point location by the use of a manually-operated search dial, an operation parallel to the analog technique in which the tape reels are manually rocked to search for the precise edit point.

By pressing the EDIT POINT button during the playback of a PCM tape, the memory circuit stores the edit point information as well as the digital signals within a six-second "window" (3 seconds before, and 3 seconds after the edit point).

When the search dial is turned, the turning speed and direction are detected and the stored data is read out and fed to the D/A input of the PCM processor. The analog output is heard through loudspeakers.

EDITING ACCURACY (RESOLUTION)

The editing accuracy of a digital audio system using a VTR has to utilize the VTR frame (1/30 sec.) as a unit. However, an audio signal requires much more accurate resolution than

Number	f_s (kHz)	m	n	f_m (MHz)	H_{PCM}
1	44.59540	248	175	249.7342	248
2	44.41558	247	175	248.7273	247
3	44.23576	246	175	247.7203	246
4	44.05594	7	5	7.048950	245
5	43.87612	244	175	245.7063	244
6	43.69630	243	175	244.6993	243
7	43.51648	242	175	243.6923	242
8	43.33666	241	175	242.6853	241
9	43.15684	48	35	48.33561	240
10	42.97702	239	175	240.6713	239
11	42.79720	34	25	34.23776	238
12	42.61738	237	175	238.6573	237
13	42.43756	236	175	237.6503	236
14	42.25774	47	35	47.32867	235
15	42.07792	234	175	235.6364	234

CANDIDATES FOR SAMPLING FREQUENCY

Figure 4. Candidates for sampling frequency.
f_S = sampling frequency
m:n = ratio of f_S to f_{EQ}
f_M = master clock frequency
H_{PCM} = H lines per field.

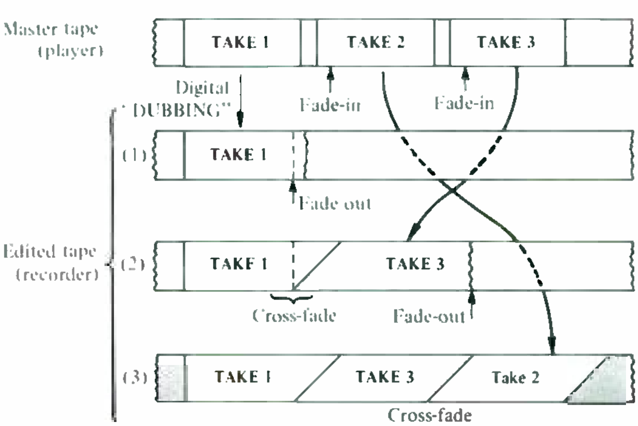
that of a VTR frame unit, and therefore, an address within a frame is required to determine at which part of the tape the digitized PCM signals are recorded. The DAE-1100, therefore, uses SMPTE time code as the reference standard for all editing controls. Since one VTR frame contains 1,470 PCM sampling data words, the edit point has to be located by detecting the corresponding frame and word through calculation of the memorized address.

The data is recorded in terms of a frame on the videotape. In some types of PCM code, one datum spreads over more than one frame. The edit point, therefore, is controlled through the digital input of the digital audio processor in sampling form of the PCM signal. The old data is read out and then re-recorded up to the edit point: the new data is recorded following the same edit point. The DAE-1100 employs a resolution level of 363 micro-seconds (equal to a 16-word PCM signal).

SIGNAL ARRANGEMENT AT EDIT POINT

Direct connection of two music signals at an edit point will cause discontinuity of signals and create noise, as shown in FIGURE 7. A crossfade, therefore, is required to produce a smooth and continuous signal and still keep the high editing accuracy. The crossfade corresponds to the slant cut used in analog tape editing. It is performed by fading out the preceeding signal and fading in the following signal. The fade-in and-out time is called the crossfade time, which can be set at ten different steps between 1 and 99 milli-seconds.

Figure 5. The digital editing process. The edited tape is created by sequentially re-recording the required segments from the original studio master tape(s).



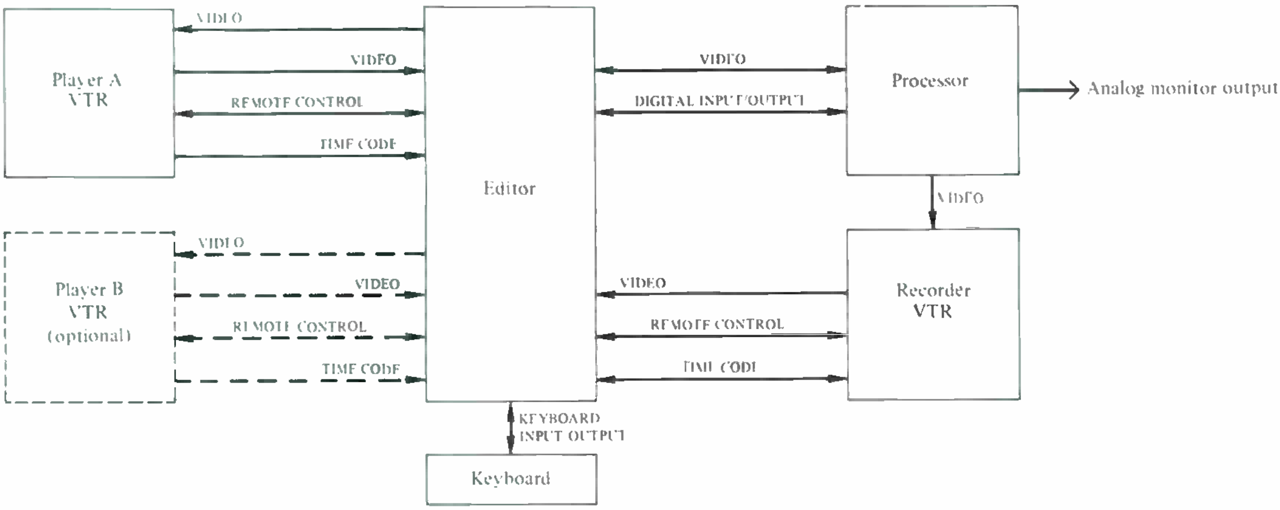


Figure 6. The complete editing system.

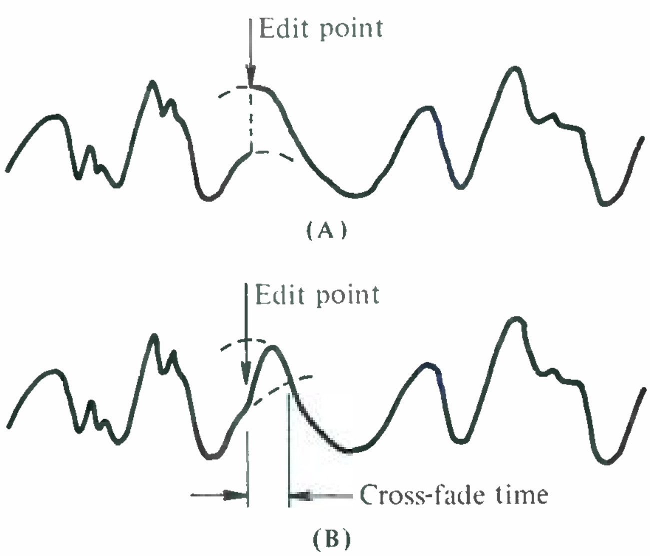
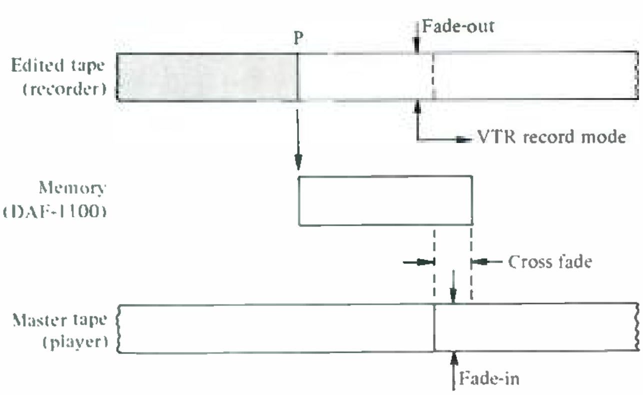


Figure 7. The crossfade process removes any audible discontinuities at the edit points.

Figure 8. The signal path during the digital editing process.



REHEARSAL AND EDITING OPERATION

One of the merits of digital editing is that the edit can be rehearsed, allowing for changes in crossfade time, level matching, and shifting of the edit point. The editing system uses only one digital audio processor, and this means that exact synchronization between the two VTRs (player and recorder) is required. In addition, as explained above, the memory circuit has to store the data necessary for the crossfade (which cannot be played back by the two VTRs during editing), plus the old data which is to be re-recorded on the edited tape.

The 16-bit, sampled data is memorized in the same memory circuits used for manual search purposes (the search function is not used during rehearsal and actual editing). The main part of the PCM data to be stored in the memory prior to rehearsal is that adjacent to the edit point on the editing tape with which the crossfade as well as the switching of playback video input of the PCM processor are controlled.

The signal paths are the same in both rehearsal and in actual editing. The only difference is whether the recorder is in the record mode (actual editing) or not (rehearsal). FIGURE 8 explains the signal path in editing.

The NTSC output of the edited tape, applied to the video input of the digital processor, is decoded into PCM signals which are then applied to the D/A input for reproduction as musical signals. Reading of the data stored in the memory (prior to the actual editing operation) starts when the edited tape reaches a point, P, six frames before the fade-out point. When this data is applied to the D/A input, the two VTRs are synchronized so that the fade-out point of the recorder and the fade-in point of the player will coincide. When the edited tape reaches the fade-out point, the synchronized PCM signal of the player is read out and the crossfade starts. After completion of the crossfade, the synchronized signal of the player continues to be reproduced as output sound.

In actual editing, when recording on the editing tape is required, the same PCM signals as those applied to the D/A input are applied to the encoder input. Since the signals for the D/A input are delayed for monitoring purposes, the input to the encoder is applied slightly earlier to realize continuous signal on the tape.

In short, continuous signal is realized by switching the signals in the following order:

Recorder's Signal → Memorized Signal → Synchronized Signal of the Player.

■

HOWARD SHERMAN

Focusing on Boston's Centel Video Complex

An insight into the thinking behind the creation of Centel.

TO APPRECIATE THE MANY facets, advantages and innovations represented by Centel (Boston), it is important to know something of Ross Cibella, the man for whom Centel was designed.

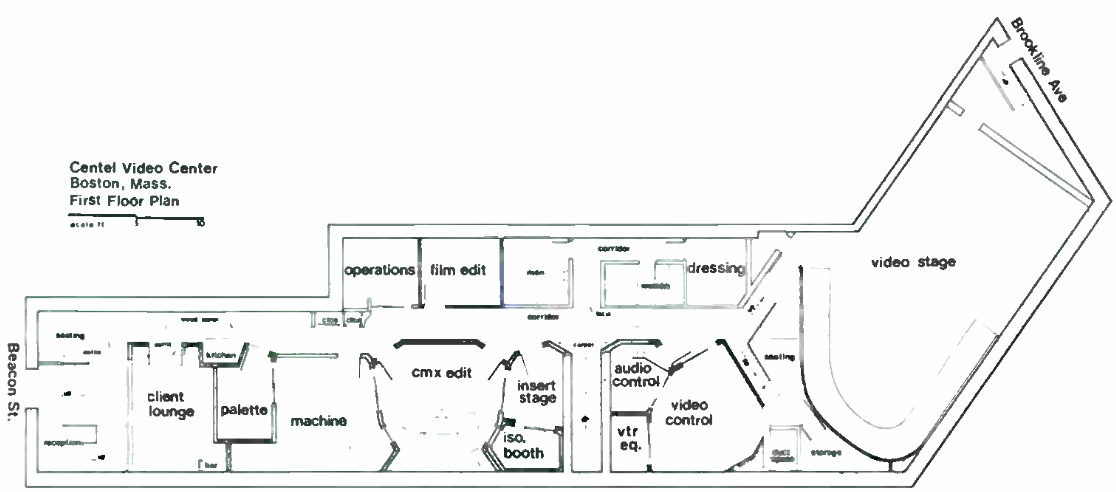
Ross Cibella is one of the most successful tv commercial directors in the Boston/New England area, a perfectionist who demands the best performance from both his people and his equipment. Several years ago, he asked John Storyk to design for him a small, 16 track audio facility, which he dubbed Century III. As Cibella explained this initial project, he was so frustrated by the then (1976) current New England studio scene, he wanted his own "state-of-the-art" operation.

Looking to create the largest 1-in. and 2-in. independent post-video production facility in New England, Cibella and Storyk teamed up again. First they had to find the appropriate building. After scouting a number of potential sites, they came up with an old RKO theatre-turned soundstage, a building in need of considerable repair. Situated on Brookline Avenue in Boston, near Fenway Park, the three-story structure represented approximately 15,000 sq. ft., and stands today as an outstanding video complex.

THE PROGRAM

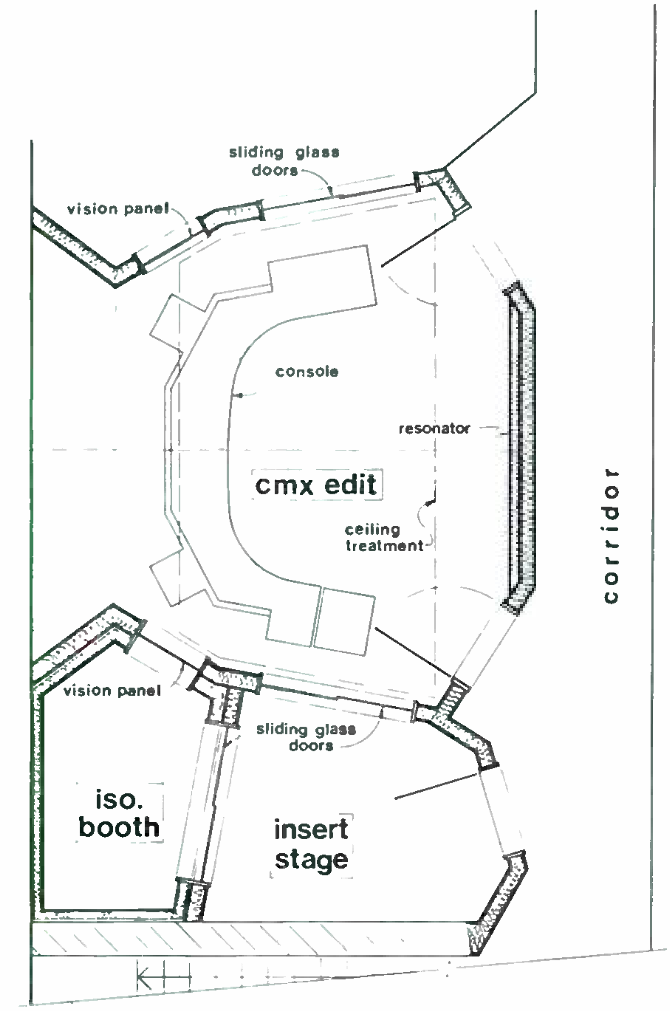
The Centel program is a remarkably sensible one. The first floor is the "core" video production and post-production complex. It includes: (1) A 30-ft. x 60-ft. video shooting stage with hard cyclorama (an area large enough to accomodate most commercial productions). (2) A video control room. (3) A

First floor plan of Centel Video Center

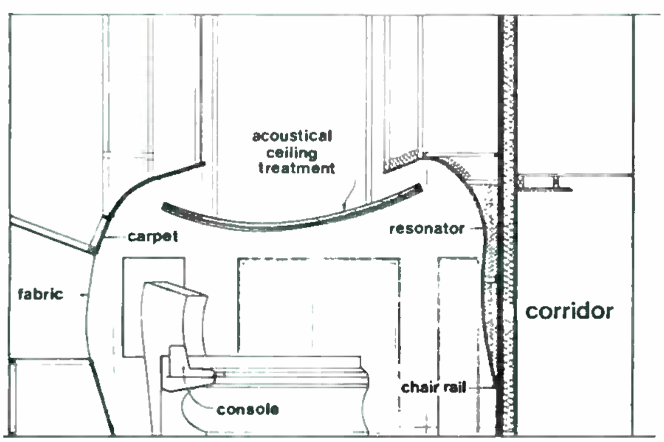


Mr. Sherman is the president of Howard Sherman Public Relations.

Plan and section of Centel.



Plan



Section Thru CMX Editing

scale ft. 5 10

The studio control room under construction.

computerized editing/post production suite. (4) An insert stage (see plans). (5) An equipment bay. The second floor provides support space: offices, graphics dept., etc.

The facility's video control room, separate equipment and audio bays, audio annex and, the cornerstone of the complex, an editing bay with CMX 340 computerized editor for both one and two inch tape (see photos and plans) make Centel New England's first computerized on-line post-production video facility.

This video editing suite is more than the true heart of the Centel complex, for it also represents a possible new generation of video studio design. Until now, the cost factor involved in equipping video studios has, with very few exceptions, prevented owners from paying adequate attention to the architecture of the rooms. Video studios were far more concerned with gear and hardware than with environment and acoustics. That the audio portion of most video productions has, historically, been an area of very little concern to video-makers is a given. This is understandable, however, when it is remembered that video sound has, until recently, been delivered to the home viewer through a 3-in. speaker.

CHANGING TECHNOLOGY

But television technology is changing. Sound is about to explode into a determining factor in the video world. Cibella, well aware of this impending revolution, wanted not only the best-sounding video post-production facility, but a facility which met pro audio specifications. According to Storyk, the single

An overview of the studio area.

biggest mistake made in designing studios is the failure to establish a room program...establishing exactly what is expected from the room. The basic concept for Centel's video editing suite was that it work for everyone, from the agency client to the video director and editor. And also, it had to meet those pro audio requirements.

Architect Storyk, client Cibella, and Centel's chief video engineer Richard Parent, spent considerable energies collaborating on the specific equipment package and physical priorities for the editing suite. The room had to be large enough to accommodate 15 people and it had to be tough, durable...a room made for long hours and heavy traffic. Curiously enough, it was not an exceptionally expensive project:

Centel was built for well under the normal $100 per sq. ft. construction costs.

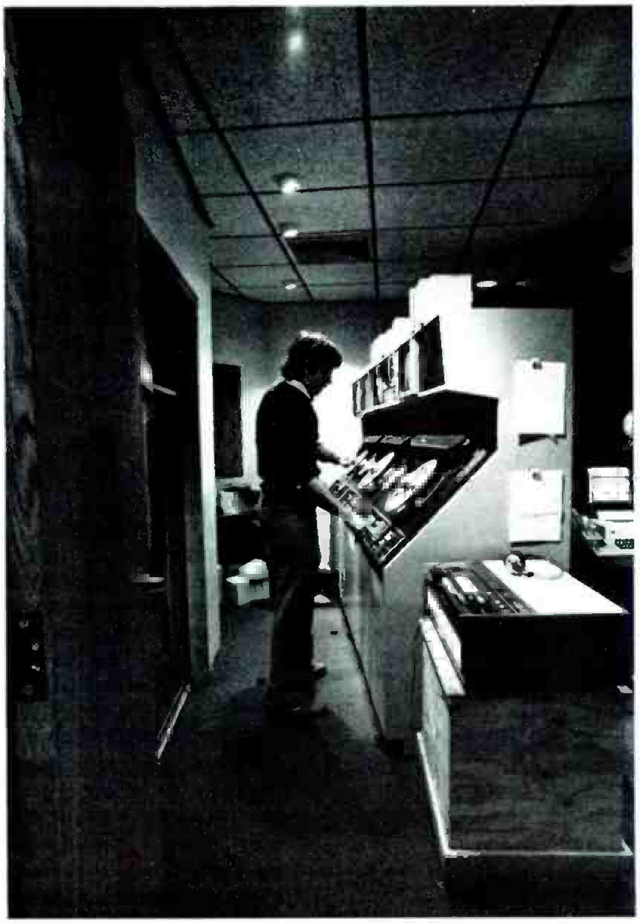
ERGONOMICS

To provide the extra physical space required by Centel, it was suggested that the video tape machines be removed from the editing suite. This is a concept which is readily accepted in the video world. (As another example of equipment isolation, see "A Studio For Research" in our February issue—Ed.)

Storyk predicts that a new cycle of audio facility is about to find widespread acceptance. Until recently, equipment manufacturers and studio owner/builders have been running on somewhat different tracks. Control rooms have been filled with expensive hardware which simply does not have to occupy that space. "How many times," Storyk asks, "have you found a $50,000 tape machine doubling as a coffee table?"

For Centel, the design program suggested that the tape machines be visible, but that there was no reason for them to intrude on valuable working space. The controls, the video consoles and the physical elbow room were priorities to be considered. The hardware went into a separate glassed-in bay.

The quad machine room.



Master Control Room, Centel Video. (Photo by Robert Wolsch Designs.)

ARCHITECTURE & ACOUSTICS

An important objective in the design of Centel was that clients be assured of basic creature comforts...task lighting, efficient climatization, complete isolation between rooms, perfect monitor visibility, etc. The acoustic and audio specifications include: (1) Room ambient of NC-25; (2) Room transfer between insert stage and mixing suite of STC-52; (3) Listening position frequeny response of ±1.5 dB (30 Hz-60 kHz) and reverberation time of 0.25 seconds at 1 kHz.

In addition to obvious eye appeal, there are serious reasons for the editing suite's physical design (see illustration). The top curve of the ceiling, a splayed, polycylindrical diffractor, serves simultaneously as a resonator and a scattering element. The concave curves focus at different points in the room and never anywhere near the listeners' ears. The convex surface curves, the single most important design element of the space, are arrived at from room-mode analysis. In addition to serving as a shell for the air ducts, wires and other gear "buried" in the ceiling, these curves work to make the suite appear to be larger than it actually is. The editing suite is only 16 feet deep by 19 feet wide, but it can accommodate 12 to 15 people, and feels roomy and comfortable.

One of Storyk's personal goals for Centel was to insure that visitors would have no sense of the orthogonal nature of the building. The interior of the structure had a "loft" feel, no windows, just a giant box. Since there was no escaping the reality of having to enter at one end and walk down a corridor to get to the other, Storyk provided an architectural texture which would make it look and feel more interesting (see accompanying plan). The jogs and nitches in the corridors work to break the possible monotony of the space.

DESIGN PHILOSOPHY

Philosophically, the Centel Complex is very much in step with most audio-only recording facilities. Centel is a factory where work is done in an artful way by people in the media world. The physical space must satisfy a combination of people, machines and tastes and, the facility had to fit within a specific set of financial and technical parameters.

The fundamental difference between audio and video studios to date has been the way artistic decisions are reached...audio decisions are made by ear; video by eye. As the audio/video marriage (revolution, explosion; call it what you will) becomes a *fait accompli*, and with every home becoming an individual self-contained "media center," audio will become an increasingly more sophisticated and important facet of all video productions.

Centel is ready now for the most complex and advanced audio/video mixing projects. It may well be a forerunner of the studios of tomorrow. ■

An A/V Update

Presenting our random access survey of a handful of hardware, as audio meets video.

NOW THAT THE AUDIO and video worlds are discovering that peaceful coexistence may not only be possible, but possibly profitable as well, we may all look forward to bigger and better A/V productions. And where will some of these productions be seen and heard? Why, on television, of all places! For it seems that some TV producers have not only discovered audio, they've discovered *digital* audio.

According to a recent 3M press release, the March 29th "Meet the Press" telecast on network TV featured digitally-recorded audio. If you were tuned in and didn't happen to notice, perhaps its because you weren't paying sufficient attention—to the *commercials.* These were produced at Sound 80 Recording Studios in Minneapolis, and 3M believes it's the first network spot to utilize digital audio. (The video was produced by Wilson-Griak, a local video production house.)

To celebrate the occasion, 3M produced its own videotape of the recording session, which will be shown at this month's convention of the Audio Engineering Society in Los Angeles (12-15 May, Los Angeles Hilton). The audio and video will be synchronized by a standard BTX-4600 controller and a 3M-designed prototype VCO interface system.

VSC LSI IC AT CES

If you find your video productions are a bit slow-paced, the VSC (Variable Speech Control) Company demonstrated "speed viewing" along with "speed listening" at the Consumer Electronics Show earlier this year. Crediting a new custom IC chip which utilizes LSI technology, VSC claims that speed listening can now be a cost-effective feature in video cassette recorders.

A VSC press release cites Videoplay Report editor Ken Winslow, who suggests that television viewers who want timely news will simply program their VCRs to record the news, which will be played back later on at double speed. In addition, videotape viewers may find these slow-paced old movies are livelier when watched at an increased playback speed.

In fact, with the proliferation of commercial, cable and pay TV, VSC's president Marvin Flaks feels that viewers truly need speed viewing/listening simply to enjoy all the shows they want, and still have time for other leisure activities.

AUDIO ON VIDEO

When the consumer decides to use his video cassette recorder for PCM audio, there's a good chance he may encounter fewer compatibility problems than now exist in pro audio. A Technical File published by the EIAJ (Electronic Industries Association of Japan) prescribes standards for a consumer-use PCM encoder-decoder, to be used in conjunction with video tape recorders. Excerpts from the standard are given here, for comparison with various pro audio digital audio systems. The numbering system is as used in the complete Technical File. The File deals mainly with VTRs in the NTSC system, which has been adopted in Japan and the United States.

1. OBJECT—This technical file is prescribed to establish the signal format and other necessary conditions for a consumer-use PCM encoder/decoder which records or reproduces audio signals in the form of pulse-code modulation, in conjunction with a consumer-use cassette video system or a part thereof, and to maintain optimum performance between the various equipment, the compatibility of recorded tapes (within the same format), and of encoder/decoders.

2. SCOPE—This technical file applies to the consumer-use PCM encoder/decoder for transforming two sets of audio signals with a frequency range below 20 kHz, in the form of pulse-code modulation which conforms to the 60 Hz/525 line television system.

3.2.1. SAMPLING FREQUENCY—The sampling frequency shall be 44.056 ±0.005 kHz.

3.2.3. QUANTIZATION—Recording shall be made in the form of a 14-bit linear slot.

3.2.4. CODING—A 2's-complement binary code shall be used. A positive binary value represents a positive audio signal voltage.

3.2.5. TRANSMISSION RATE—The transmission rate shall be 2.643 Mbit/sec.

3.3 PCM SIGNAL FORMAT—The PCM signal format shall be in accordance with television standards; one field (262.5 H) contains a data block of 245 H, and a control signal block of 1 H.

3.3.2. DATA BLOCK—One data block shall consist of six sampled words, P and Q error-correcting words, and one error-detecting word (CRCC). The data block of nine words has a span of 1 H.

3.3.4. SYNCHRONIZING SIGNAL—Horizontal and vertical synchronizing signal formation shall be in accordance with the 60 Hz/525 line television system.

3.4.1. SAMPLED SIGNAL WORD—The sampled signal word shall consist of 14 bits, which are laid out in such a way that bit 1 is the MSB (most-significant bit) and bit 14 is the LSB (least-significant bit).

HI-FI PCM

Superficially resembling a video cassette recorder, the Technics SV-P100 Digital Audio Cassette Recorder combines

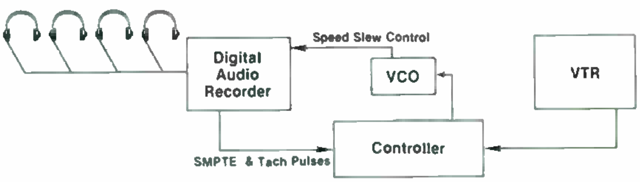


Figure 1. Using a BTX-4600 controller and a custom-designed VCO interface system, 3M provides digital audio for video productions. (A) the 3M TT-7000 video tape recorder. (B) System block diagram. (C) 3M digital tape recorder.

the helical-scan NTSC system with a PCM audio processor in a single unit. The system conforms to the standard listed in the EIAJ Technical File.

In addition to the usual record/playback features, the SV-P100 offers "jump" and "search" functions. When the user does not wish to listen to certain portions of a tape, these may be skipped automatically, at eight-times normal speed. To do this, a "jump mark" is recorded on one of the auxiliary tracks during the appropriate portion of the tape. When the deck encounters the jump mark during playback, it immediately skips to the end of the jump mark, where it resumes normal playback. Jump marks may be erased by pressing the "clear" key.

Search marks may also be recorded on the tape. During rewind or fast forward, depressing the search key will cause the deck to seek out the nearest search mark. During the search, if the playback key is depressed, the machine will automatically go into the play mode once the search mark has been located.

UPGRADING SLIDE-SHOW AUDIO

Even the 35 mm slide show has undergone remarkable changes over the last few years. Some 20 years ago, the 50/30 (Hz) slidefilm projector was the industry workhorse, with the 30 Hz tone triggering the slide advance. But as Martin Dickstein has often pointed out in our "Sound with Images" column, today's slide show is often a multi-projector, multi-screen production, which demands high-quality audio to complement the sophisticated video.

Neal Ferrograph's model 330 audio/visual cassette recorder is aimed directly at the A/V pro who requires state-of-the-art (cassette) audio with his images. The model 330 is a *three*-channel system, designed for easy interface with most 35 mm slide projectors.

Although all three channels feature full audio-frequency bandwidth, the system is designed for two-channel audio, plus sync applications, and Dolby B-type noise reduction is provided on the two audio channels only. Of course, when sync is not required, the third channel may be used for non-Dolby audio if necessary.

Simultaneous or independent operation of the sync channel is possible. For single-projector control, a built-in 1000 Hz sync oscillator records a 450 ms pulse on the third channel whenever the front-panel pulse button is depressed. On playback, an internal relay senses the recorded pulses, and provides a contact closure to operate the projector's slide-change mechanism. For multi-projector applications, the internal relay may be switched off, and the user's own (external) projector-control signals may be recorded onto the sync channel.

Figure 2. Intelligible audio allows the VCR owner to experience high speed viewing/listening. Shown here is VSC's remote control unit.

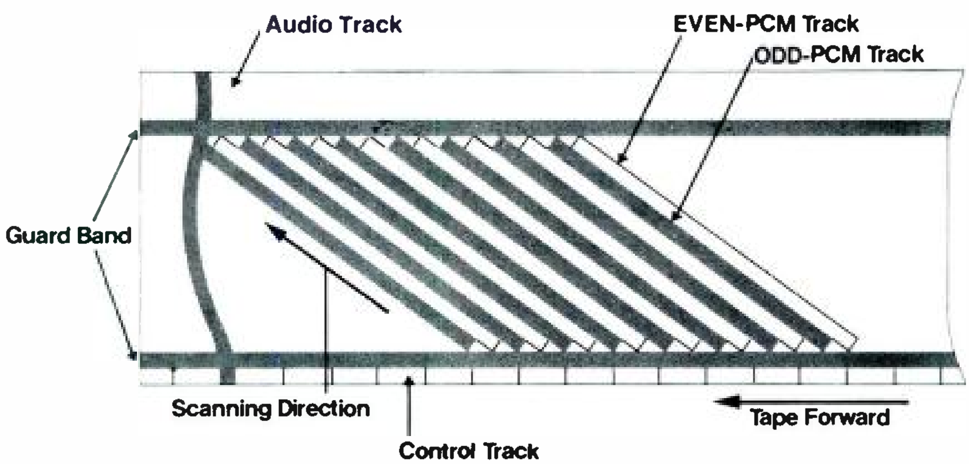


Figure 3. PCM audio recording on a VHS format video cassette recorder.

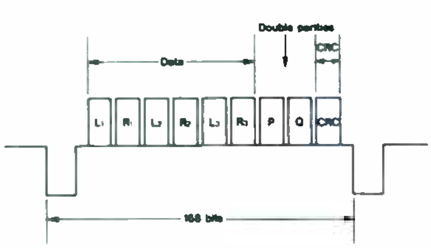


Figure 4A. Signal distribution on one horizontal scanning line (1 H). Instead of a TV video signal, three samples of the quantized audio signal are recorded, along with double-parity error correction and CRC checks for drop-out detection.

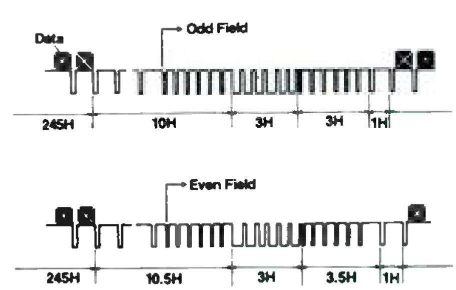


Figure 4B. After every 245 H lines, horizontal and vertical sync pulses are inserted. The resulting format corresponds to NTSC TV standards.

Figure 5. The Technics SV-P100 digital audio cassette recorder.

Figure 6. Panasonic video equipment.

A/V TO GO

In more cases than not, the "video" in the typical A/V production consists of 35 mm slides, most-often found in Kodak carousel trays. As noted above, audio cassette recorders such as the model 330 provide the capability for high-quality audio.

But what about eventually replacing those slides with video tape? With unlimited funds, anything is possible, but for producers on a budget, those production costs are one reason why slide shows retain their popularity.

With the development of relatively low-priced video cameras, all that may change. There was a time when the average producer couldn't even think about owing his own camera. Now, you can buy a reasonably good one for about $1,000.

To find out just what to expect from such a camera, we recently borrowed a Panasonic PK-800 color video camera, and a PV-3100 portable video cassette recorder. By a remarkable coincidence (?), the equipment arrived just before we left for a two-week trip to South America. To make a long story short, the gear was subjected to two weeks of equatorial sun, ocean spray, jungle dust and humidity, and altitudes from sea level to 12,000-plus feet. Although surely not designed with this kind of abuse in mind, it stood up admirably (sometimes, better than its operator).

Back home in the comparative safety of our lab, our video color monitor shows us that we have many hours of excellent video tape ready to be edited. Our mono-only audio is certainly not "hi-fi," but it's not bad either. However, it would be nice to have stereo record capability, since the problems of tying up to an auxiliary two-channel audio cassette recorder are really not practical. (Remember, this is a budget production.)

Our efforts will certainly not be picked up by the National Geographic Society for prime-time viewing. In fact, it probably won't even make it to cable TV. But it was a rewarding experience for someone who was previously having trouble enough trying to cope with audio.

We'd highly recommend the use of this, or similar equipment to anyone interested in getting their feet wet (in our case, literally) in video production. For the audio pro who's not quite ready to go full-speed ahead into video, it's a reasonably painless way to get some experience in the medium. ■

JOHN BORWICK

The 68th AES Convention

From the Congress Centrum *in Hamburg, John Borwick reports on the tenth European convention of the Audio Engineering Society.*

HAMBURG, GERMANY: Visitors to the 68th AES Convention (March 17-20) were left in no doubt that the audio industry is now largely preoccupied, almost mesmerized, with digital techniques. Around one-third of the 45 scheduled papers related to digital, as did several out-of-hours sessions, such as the crowded meeting of the AES Digital Audio Technical Committee, and a number of get-togethers of an *ad hoc* committee formed to discuss Input Output Digital Interface Techniques.

Even people not yet committed to digital equipment in the marketplace (whatever their backroom research status) had plenty to say about digital. In addition, the downright anti-digital school were also vocal on the subject. Dr. Ray Dolby, this year's AES President, was quoted in a trade paper as saying that, "Digital, as it applies to most problems of the workaday world, is pretty much technical overkill." I also heard much talk, but no positive evidence, of "digital tape death" (a fatal condition said to afflict digital tapes when the frequent passes needed during editing cause such serious degradation that the error correction system cannot cope). Alastair Heaslett of Ampex gave an evening talk on some other factors which, it seemed to him, made the various digital recorders already available of doubtful cost-effectiveness. His company were also demonstrating a special version of their ATR 104 analog tape recorder with two extra-wide tracks on half-inch tape. Running at 15 ips with no voice reduction, this certainly sounded pretty good—and devoid of tape hiss. To show that they felt this standard of recording was well up to the requirements of current digital equipment, they handed me an A B switch to let me compare the direct off-tape sound with that fed in and out of a typical analog-to-digital-to-analog processor, either simultaneously or with delay. I must admit that the sound out of the processor was less than the "super-analog" sound going in.

Other recorder manufacturers are also looking at super-analog. The French firm Enertec-Schlumberger, for instance, were showing a "Grand Master" version of their F462 tape recorder, adapted for two tracks on half-inch tape, running at 15 or 30 ips. They were promoting this machine as suitable for studios that intend to switch to digital audio discs. It's quite a twist when so many recordings are made on digital masters and then transcribed to conventional analog LPs, while the promised excellence of future digital *discs* is used as an argument for super-analog taping. This would of course keep studio costs down and allow traditional mixing and editing techniques to continue.

John Borwick is **db***'s British correspondent.*

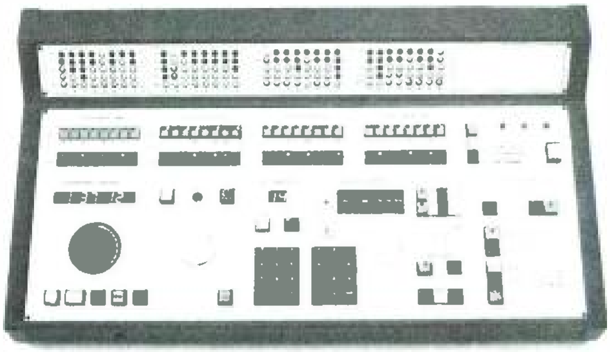


Figure 1. For digital recording, the EMT-450 Digiphon control system is used in conjunction with a CDC Data Storage System. The CDC uses a stack of hard magnetic discs.

Figure 2. JVC's latest portable PCM audio processor, for use with videocassette recorders.

DIGITAL RECORDERS

Of course, there was the full parade of available digital recorders. Most novel was one which, instead of tape, used a stack of hard magnetic discs. Advantages claimed were the rapid access time which discs provide for cueing and editing, and the fact that the data storage system itself is already being marketed widely by CDC, and so engineering back-up is available. To adapt this 300-Megabyte storage medium for digital audio recording, EMT-Franz designed their EMT 450 Digiphon control console. The disc drive has such a high data transmission rate that up to 32 tracks can be recorded simultaneously. The 16-bit, 32 kHz sampling frequency system gives a total storage capacity of 140 minutes for mono, with proportionately shorter durations for two or more tracks. The signals are fed into a buffer store in block form with all access or processing performed in the sync gaps. I was given a demonstration by Karl Bader of EMT-Franz, who also read a paper on the subject on behalf of his co-authors Barry Blesser and R. Zaorski. Access time to any required address was a matter of milliseconds, and the sequence to carry out a normal edit was similarly speedy: one simply identifies "exit," "enter" and "go to" points. A touch-button allocates address flags to these points, whereupon the edit can be rehearsed and fine-tuned by another control which shifts the flags in 5-millisecond steps as one listens to the join being recycled over a period of, say, 5 seconds.

JFC (Victor Company of Japan), besides their main studio PCM recorder and AE-90 editor, introduced a neat portable PCM Audio Processor. This was switchable to PAL SECAM (44.1 kHz) sampling frequency) or NTSC (44.056 kHz) to match both types of videotape recorder. It was a two-channel, 14-bits linear device and was being shown with JVC's equally compact HR-2200 VHS recorder, weighing only 5.2 kg (11.4 lb) including battery pack.

Mitsubishi were also unveiling a prototype of the X-800; a 32-channel update of their original X-80 2-channel quarter-inch PCM tape machine, which has already found a fair number of customers worldwide. The X-800 uses 1-inch tape running at 30 ips. All 32 channels are available at all times and separate tracks take care of SMPTE coding, sync, error correction and audio analog signals. Though Mitsubishi claim the X-800 is immediately operable by anyone familiar with multitrack analog, the built-in microprocessor provides numerous automatic and memory functions such as "ping-ponging" (frequent mix-down routines while relocating tracks), sync recording (a digital buffer memory ties record and playback signals to within microseconds), punch in and out, variable ptich (±10%). The 44 heads are assembled in-line for easy replacement and realignment.

3M's Mincom Division were showing their Digital Mastering System comprising 32-track one-inch, and 4-track half-inch machines, with a flexible remote control unit having five LED level indicators for each track. The 3M Digital Editor and

Digital Preview Unit were also attracting plenty of attention. The system introduces delays of up to 1.3 seconds in 5-millisecond steps, to provide a preview signal for cutting-lathe pitch and depth control. About 40 albums and 20 singles ha· · already used the 3M system, featuring artists from Herb Al· and Chicago, to Herbert von Karajan and the P 'in , harmonic Orchestra.

The Sony DAF-1100 Digital Audio Editor has been seen at shows before, where it is always played with by budding digital engineers. Here in Hamburg, they added a new Digital Audio Processor (PCM-1610), interchangeable with the familiar PCM-100, but with a higher spec and ready for instant interfacing with professional VTRs.

The big news from Soundstream, recently acquired by the Digital Recording Corporation (DRC), was their plan for extension into Europe. Instead of negotiating for the expensive hire of Soundstream machines shipped from the USA · nd flying to Utah to carry out the editing—Old-World producers can now negotiate direct with a new DRC-Soundstream digital editing facility being set up in Gutersloh near Hannover, West Germany by negotiation with the prestigious disc pressing company Sonopress. DRC Soundstream have been building more of Dr. Tom Stockham's digital recorders—which were busy at AES Hamburg demonstrating JBL loudspeakers in one suite and the Soundsteam recorded repertoire in another. The company is also working on an editing facility for Los Angeles and one in London within six months. Forthcoming DRC Soundstream plans include a consumer digital audio player using cards made of film instead of the different disc systems previewed so far by Philips, Telefunken, JVC *et al.*

MIXING CONSOLES

Of course, digital techniques do not stop at the recorder stage. They are also revolutionising our ideas about mixing consoles and signal processors of all kinds. Though US-based console manufacturers were showing in strength at AES Hamburg—notably Harrison, MCI and 3M—it is the European firms who naturally tend to introduce new products at the European Convention.

About the biggest desk shown by anybody was a 56-channel 48-track Neve 8108 monster which had to be carried in through the Conference Centre kitchens and restaurant, as it could not get along the corridors. Neve's engineers were happy to tell me that this desk was not going back to England but would be delivered to a German customer right after the Show. They had also made a special feature of a simulated TV and film dubbing suite, complete with videorecorders, synced 24-track, cartridge effects players, etc. This was to demonstrate the versatility of their NECAM II computer-assisted mixing system, which I had recently seen in action at the BBC Television Centre in London.

Figure 3. From Hungary, Elektroimpex introduced its FIT-IC automated console system.

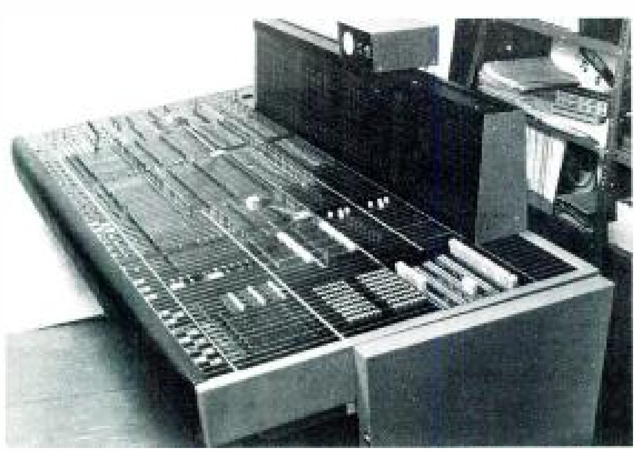


Figure 4. The SAM 4_ , _ _le mixer from Sweden's Satt Electronics

Solid State Logic were also proud of th_ _ BBC connection, having recently installed a 32-channel version of their 4000E "Total Recall" automation system in th_ _ _ _'s newest music studio, the Concert Hall in Manches_ _ . _heir client list numbers nearly 40 studios, of which 13 are in the US _ _ latest version of this system, shown in Hamburg, besides tying each control above the channel faders directly to a computer to memorize complete status details, had a new comprehensive bar-graph with legends above each channel which could be switched to act as a real-time analyzer display of the response of any chosen signal.

Electroimpex of Hungary were newcomers to automation with the BEAG automated console, designed particularly for broadcasters. Up to 60 1-inch width in-line modules can be accommodated, with 30 module options. All auxiliaries can be integrated with the main frame and the version on show had a particularly comprehensive LED display, with real-time analyzer.

I was intrigued to see that the TEAC Corporation had introduced automation on a prototype addition to their Tascam range of semi-pro desks. They played me a recorded sequence to let me see how rows of LEDs alongside each fader gave continuous indication of the fader status with time, while a circle of LEDs round each pan-pot similarly showed the left/right panning status: ingenious. Another new exhibitor was Mondial Electronique of France. In anticipation of a big expansion of local broadcasting in France (something we are also witnessing in Britain), they had collaborated with Radio France in developing new radio studio equipment. The PR14-ME/S presenter's console was particularly interesting. Rows of buttons alongside the two-channel faders allowed for the selection of up to 10 stereo sources. Up to three sources can be used at a time, with remote cueing and start of disc or tape players and a programmable timing device to give sequential operation.

SATT Electronics of Sweden showed small portable mixers with interesting specifications. Their SAM 42 (around $3,000), measuring only 11.3 x 10.4 x 3.1 inches, had 4 mic/line inputs, 2 monitoring inputs, 2 master outputs, 1 auxiliary output, built-in talkback microphone, test oscillator, peak programme meters and phantom mic powering.

FOR THE REST

Kudelski of Switzerland were still not ready with their long-awaited Nagra T-Audio desk-size analog recorder. However, they did have preliminary leaflets and the next AES Convention should let us see the machine itself. This will be a dual-capstan design, with the Nagra pre-distortion technique giving up to 6 dB better signal-to-noise ratio. The detachable keyboard operates on a serial digital bus, allowing one machine to be operated from several keyboards built into mixing consoles. Spools of up to 12-inch diameter of ¼-inch tape can be used at all four normal speeds, 30, 15, 7.5 and 3¾ ips with ±7 percent varispeed control.

Tandberg of Norway have been advocating a change to a lower equalization time constant (10 microseconds for 15 ips, 25 μs for 7.5 ips) and claiming a boost to 80 dB signal-to-noise ratio in the process. They showed several versions of their TD20A-SE open-reel tape deck using this idea and it certainly produced inaudible tape noise.

Lyrec of Denmark claim that their TR-532 24-track recorder is the most complete, yet compact, multitrack recorder on the market. The machine they demonstrated to me was on its way to their 40th customer in Germany and they said there were 32 already in use in Britain. The Lyrec ATC (Audio and Tape Controller) certainly is versatile. Its autolocator is referred to the minutes-and-seconds tape timer. Up to 32 tape locations can be entered into the memory via the keyboard and searched and transferred at will. Six color-coded rows of 24 buttons control the audio tracks individually and inset LEDs show the current status—playback, sync, ready, record and solo.

Audio & Design's latest product was the Transdynamic Broadcast Processor. This split the signal into three user-adjustable frequency bands with 6 or 12 dB-per-octave phase-compensated filtering. Separate processing in the three stereo bands permits a greater degree of level compression than usual, and subjectively louder broadcast signals, before side effects become too obvious. A built-in pink noise generator simplifies calibration and line-up. Price in the USA is $4,000-$7,000, depending on the associated level control amplifier form Audio & Design's Scamp, Compex, Express or Easyrider ranges.

This interest in radio broadcasting was reflected in three different telephone devices which I noticed. Tandberg showed the TES Telephone Line Quality Enhancement System. This converts the frequency range of 100—3,100 Hz to 300—3,300 Hz for transmission from the remote end of the line. The receiver converts this back to achieve 1½ octaves more bandwidth at the low end of the spectrum for a small sacrifice at the high end. If two telephone lines can be used, a total bandwidth of 6 kHz is possible, with compression/expansion giving about 15 dB better signal-to-noise ratio. Studer also had a Telephone Hybrid unit which gives better match of telephone caller and studio voice by attenuating side-tone effects as soon as voice modulation occurs. Tore Seem of Norway showed a suitcase Commentator's Telephone package. This comprised a 2-channel mixer and AGC (automatic gain control) amplifiers with flexible telephone dialing, incoming call buzzer and lamp indicators and headset or headphone operation. The commentator simply connects to a 2-wire telephone line (by prior arrangement) and has full broadcast and monitor facilities.

Figure 5. A Commentator's Telephone System introduced by the Norwegian firm of Tore Seem.

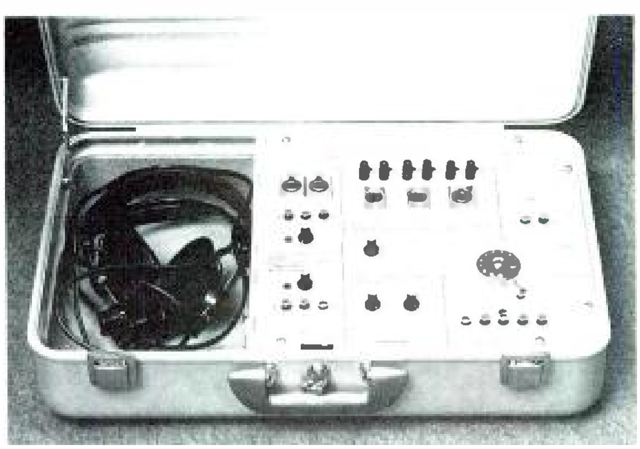


Figure 6. Beyer's SM 85 is a universal radio microphone transmitter, which will accept a variety of plug-in microphone capsules.

Schoeps were giving a persuasive demonstration of the use of their Colette miniature microphone (hyper-cardioid pattern) as a substitute for a gun microphone in TV or film applications. Of course, the directivity pattern is less narrow than that of the shotgun mic, but is is more consistent over the frequency spectrum and so the residual off-axis studio noise sounds natural and does not require the sharp bass cut used with shotgun mics. The newest idea from Beyer was the SM 85 universal radio microphone transmitter. This hand-held tube, with pigtail aerial wire, could accept any one of a wide range of Beyer plug-in microphone capsules, including condenser types, to give wireless communication with the standard Beyer receivers. Suitability for the differing needs of singers, commentators and TV floor managers was thus achieved. Sennheiser introduced a new generation of condenser microphones with a claimed extension of dynamic range. The S40 cardioid, first of the line, is said to have the needs of digital recording in mind, with low distortion and a maximum input level of 134 dB.

A switch to higher quality pre-recorded cassette duplication was very much in the air. BASF told me that several European major record companies were ordering their chrome cassette tape for future title issues, in the manner initiated by Columbia (CBS) in the USA last fall. Cetec International and King Instruments had full-scale working exhibits of their latest high-speed duplicating and cassette loading systems respectively. Infonics played me some duplicated recordings made in-cassette at 10 times normal running speed without noise reduction. Frequency response went all the way up to 20 kHz, as confirmed by a continuous real-time analyzer display, and tape hiss was remarkably low.

CONCLUSIONS

Reacting to the need for change seems to be the name of the game in today's audio industry. A visit to an AES Convention is perhaps the best way to get an over-view of this reaction. Space has allowd me to mention just a few of the newer products and attitudes. Except that attendance figures seemed a little down this year, this 10th annual European AES Convention, which dovetails between the East and West Coast US conventions, was clear evidence of healthy inventiveness in the audio laboratories of the world and adaptability in the studios. ■

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People/Places/Happenings

• **Stanley W. Faught** has been named general manager of the **Ampex** Magnetic Tape Division. The announcement was made by **Charles A. Steinberg**, executive vice president of Ampex Corporation. Steinberg, in addition to his other responsibilities, had been serving as the division's acting general manager. Faught joins MTD after a six-year career at Ampex's Colorado Springs audio-video systems manufacturing facility. During the past four years he served as plant manager for the Colorado Springs operation with responsibility for manufacturing in Colorado Springs, Juarez, Taiwan, and most recently Cupertino, California. Faught had been with **Honeywell, Inc.** for 11 years before joining Ampex.

• **Tom Beckmen**, president of **Roland-Corp US**, has announced the appointment of **David Hadler** as director of sales for the newly formed **Roland Studio Systems Inc.** Mr. Hadler comes to Roland Studio Systems with extensive professional experience. His background not only includes years of experience doing live P.A. for such groups as the **Doobie Brothers**, **J. Geils Band**, and **Traffic**, but also includes professional equipment sales, and most currently, the position of national sales manager at **Quad Eight Electronics**.

• **McMartin Industries, Inc.**, of Omaha, Nebraska, has established a regional liaison office in London, England. **Norman A. Broad** has been named regional sales manager for Europe, Africa, and the Middle East, and will coordinate all aspects of sales, service, and project management in the region. Mr. Broad has been technical director for **Allied Broadcast Systems in England**, and previously held several engineering and management positions in the broadcast industry over the past twenty years.

• The **Artisan Recorders'** Mobile Unit recently completed the first successful remote digital recording in Florida, using the **Mitsubishi** Digital Audio Systems X-80 PCM recorder to capture the **Fort Lauderdale Symphony Orchestra** live at the Fort Lauderdale War Memorial Auditorium. **Peter Yianilos** and **Richard Hilton** engineered The Artisan Mobile Unit also recently recorded **Wendell Adkins'** third live album at **Whiskey River** in Fort Lauderdale. **Peter Archer** produced the album with Peter Yianilos and Richard Hilton engineering.

• **Argos Sound** has announced that as of January 1, 1981, all amplifiers used in Argos portable sound equipment are being designed and manufactured in the Argos plant in Genoa, Illinois. This new manufacturing capability is a direct result of the appointment of **Loren Schrader** as chief electronics engineer and the establishment of an electronic product department. Argos manufactures a line of portable sound, baffles, baffle/speaker combos, sound columns, outdoor speakers and low cost total sound systems.

• **Ed Sternbach** has joined the professional audio/video and parts division of **Harvey** as its buyer. Prior to joining Harvey's, Sternbach was a buyer for the **National Broadcasting Company**.

• **Mark S. Fowler**, 39, a Republican lawyer who specializes in communications law, has been nominated by **President Reagan** to the **FCC** and, ultimately, to become its next chairman. Fowler has been named to take Commissioner **James Quello's** seat, a six-year term ending June 30, 1987. Quello is expected to stay on at the FCC and will take the seat vacated by **Tyrone Brown**. Fowler served in the Reagan transition team and as co-director of the Legal and Administrative Agencies Group, which included the FCC team. He also served as the communications counsel for the Reagan campaign.

• **Shure Brothers Inc.**, Evanston, Illinois, has announced the promotion of **Michael K. Solomon** to the position of marketing product manager for distributor microphones, with responsibilities for all distributor and private label microphone programs involving both domestic and export distributor markets. Solomon joined Shure in 1979, and prior to this promotion was technical coordinator.

• Three appointments to **3M's** Magnetic Audio/Video Products Division product evaluation and field service staffs have been announced by **Dr. John Holm**, manager technical service and product maintenance for the division. The three appointees are **Dr. Gerald S. Anderson**, manager of product evaluation and test development; **Delos "Del" A. Eilers**, manager of audio field service; and **H. Lee Marks**, manager of video field services.

• A television commercial produced at **Sound 80 Studios**, Minneapolis, is believed to be the first network spot to utilize digital audio. The corporate ad, produced for **Archer Daniels Midland**, was aired March 29th on "Meet the Press," according to ADM's advertising agency, Minneapolis-based **Martin/Williams**. Video for the 90-second spot was produced by **Wilson-Griak**, a local video production house. A videotape documentary of the recording session, entitled "The Day It Came Together," will be shown by 3M at the upcoming **NAB** and **AES** trade shows. The documentary will be synchronized by a standard **SMPTE** controller and by a prototype interface device (VCO) developed by 3M.

• **Stewart Greenberg** has been elected vice president of Marketing and Sales at **James B. Lansing Sound, Inc.**, effective March 30. Making the announcement was **Jerry Kalov**, JBL President. Greenberg will direct domestic and international sales and marketing efforts for JBL's high fidelity and professional product lines. Greenberg has a ten-year career in the high fidelity industry, most recently serving as Audio Division vice president for **Yamaha International Corporation**. Prior to assuming that post, Greenberg held a variety of sales and marketing management positions with Yamaha, and from 1971-1973 was product manager for **Fisher Radio** in New York.

• The **CBS Television Network**, a division of CBS, Inc., has selected **Rupert Neve Incorporated** of Bethel, Connecticut, as their primary supplier of sophisticated TV sound production consoles for their New York City broadcast center. An initial quantity of three consoles has already been ordered, and CBS holds options to purchase another five during 1981 and 1982. These consoles are of the new standard Neve design to specification 7056 incorporating microprocessor control of input to output routing and the ability to store in internal solid state memories central routing assignments. The console has 36 inputs, 8 submasters and 16 masters and is priced at around $300,000. Neve has long experience with the supply of TV audio consoles to CBS. The first Neve console, a model 8014, with 16 inputs and 4 outputs was supplied to CBS in 1972. Three additional large TV audio consoles were supplied to CBS TV City in Hollywood in 1976 and a substantial number of smaller to medium size consoles have been delivered to CBS over the past few years.

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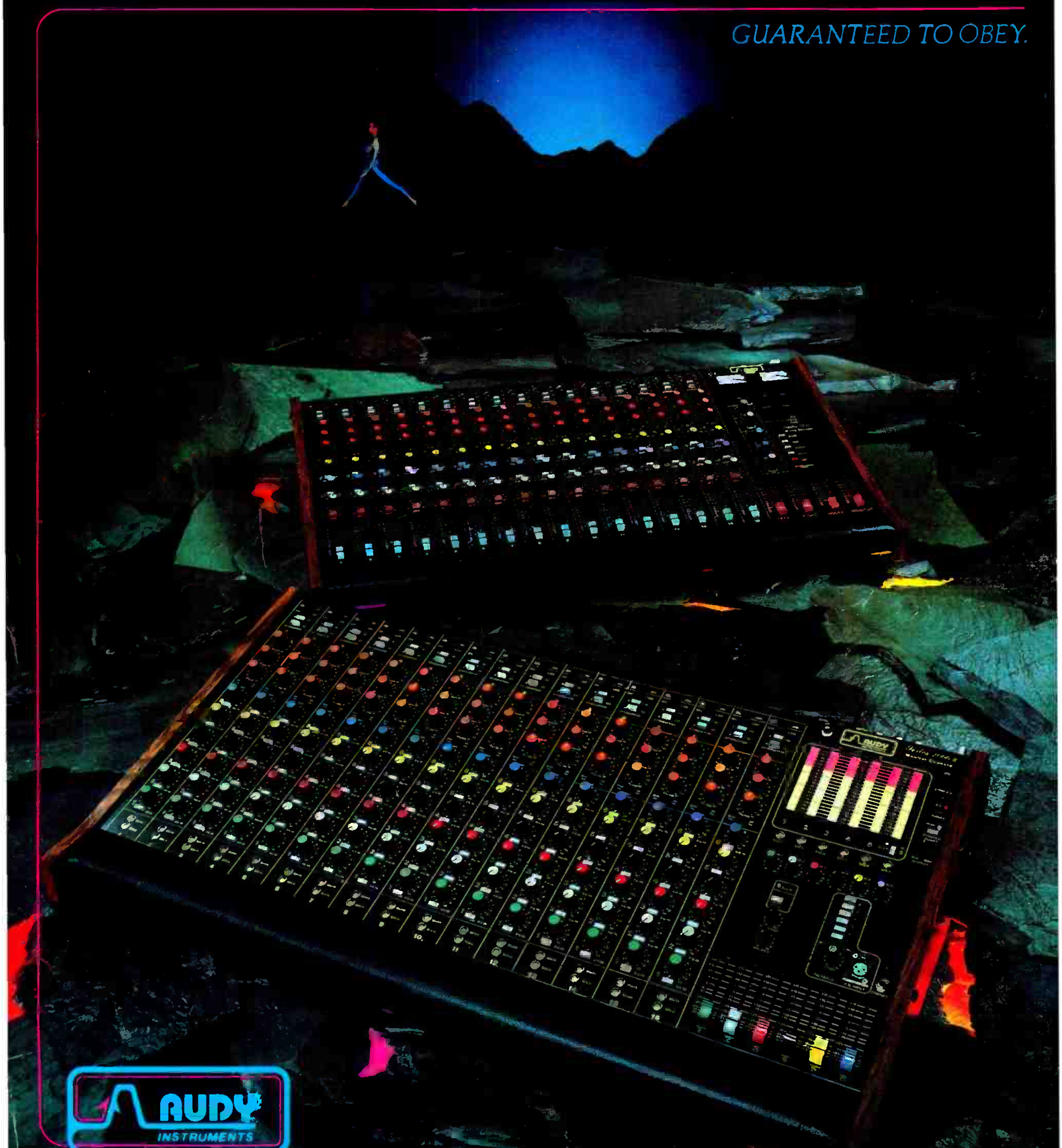

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