THE SOUND ENGINEERING MAGAZINE
MARCH 1981
$1.95

db

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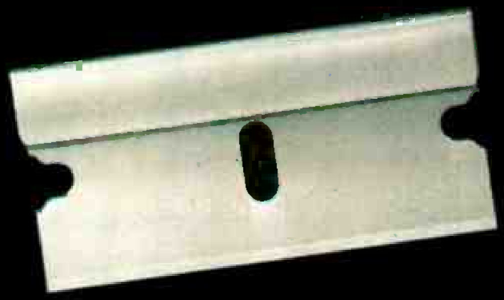

All other digital 2-channel audio recorders require special editing equipment which could raise your initial investment by as much as $25,000 or more. The Mitsubishi X80 Digital 2-channel Audio Recorders require only a standard razor blade to edit, even within the music portion.

In designing the Mitsubishi X80 series digital audio recorders, we've made the human engineering process as simple as a standard analog system but, recording with the X80 captures and plays back true sound. The depth. The presence. The richness. We call it Three-Dimensional Sound.

When you hear music captured on the Mitsubishi X80, it's like being right in the booth with the players. There's no hiss. No print through. No drop outs. No deterioration through generations. The Mitsubishi X80 (portable) and the X80A (console) are the culmination of years of refining and perfection and are available for delivery right now.

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The Mitsubishi X80 series was designed with you in mind.

Who needs digital?
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Digital audio recording is not a fad, but rather the inevitable future of recorded sound.

The Digital Audio Disk Standardization Committee, in Tokyo, has put its seal of approval on a compact disk format for home digital audio recorders that uses optical laser technology to play-back digital audio disks. Through this laser signal, sound can be reproduced much cleaner and more realistically than ever before.

Digital audio recording is as big a sound breakthrough to analog recording as stereo recording was to the Hi-fi industry. Digital audio disks and software will soon be a reality in Japan and will be available in the United States in 1982.

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X80 X80A

2-Channel Digital Audio Recorder

MITSUBISHI
DIGITAL AUDIO SYSTEMS

Circle 10 on Reader Service Card

ABOUT THE COVER—

• In the Glendale, California, R & D studios of Yamaha International, a mostly-electronic selection of the company's MI products introduces us to this month's topic of "Music in the Studio: Electronic." Reading clockwise from lower-left foreground: CP80 Electric Piano, CS40M Programmable Memory Synthesizer, B100-115SE Bass Amp, BB-800 Bass, C-7 Piano, SBG-2000 Guitar, G100-112 Guitar Amp, 9000 series Drums (in reflection), GS-1 Digital Synthesizer, and GS-1 Programmer. The programmer has four CRT displays to simultaneously represent its four oscillator banks. Photo courtesy of Thom Elder.

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THE SOUND ENGINEERING MAGAZINE
MARCH 1981 NUMBER 15, VOLUME 3

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db Letters

LOOKING FOR GARRON

TO THE EDITOR:

We write to you as a last resort in a two-year attempt to locate something very important to us.

We are, as our name suggests, a Hospital Radio Service which broadcasts light entertainment to the patients in four hospitals in the London area. We are staffed entirely by unpaid volunteers and are "on the air" (though we go out on closed-circuit) from 8:00 am to 11.00 pm seven days a week. We are entirely funded by ourselves, by subscription and donations.

Anyone in radio will tell you that a vital part of a studio's gear is the NAB cartridge machine, but these cost around £1000 in the UK and we could never afford to purchase one. However, a professional recording studio sold us one which was quite old and in need of repair, about two years ago for £50.

We write to ask if you could use your good offices to try and find a circuit diagramme or manual for this machine amongst your readership. The machine is a Garron Electronics RAPID Q. It has a record facility (this is one of the functions which has failed) and is an instant-start, subliminal-tone-cue machine.

We have been unable to trace Garron Electronics (it looks as if they've gone out of business) and no one who has had their machines has a manual left—even the BBC who used to use them years ago.

We desperately need a working Cart Machine which can record, and having spent nearly £100 which we can ill afford on this one, it seems a shame to give up without a fight and scrap it.

Index of Advertisers

Coming Next Month

• In April, it's "Music in the Studio: Acoustic," in which we continue our examination of the musical side of the recording industry (it's not *all* electronic you know). Plus more on the analog digital controversy, and a little broadcast audio thrown in to mark the upcoming NAB convention.

PRO MASTER™: the uncomplicated way to professional sound

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Circle 11 on Reader Service Card

I do hope that you can help us, and perhaps if you know any company who has taken over Garron's business, you will let us know so that we can try yet another tack. Perhaps you are aware of another broadcasting service in the USA who has used these machines who may be able to lay their hands on an old manual.

We look forward to hearing from you.

With many thanks, and in anticipation of your valued help, we remain,

David Garfield
Station General Manager
London Hospitals' Broadcasting

db replies:

How about it readers? Can anyone help Mr. Garfield?

WHAT'S A DYNAMIC CONDENSER MICROPHONE?

To The Editor:

I would like to correct some misprints in my article, "Hum, Pop, Thump, and Other Microphone Noises" in the December, 1980 issue of **db**.

In Figures 7 and 9, the frequency scale shown is ten times too large; that is, 1 kHz should be 100 Hz, 100 Hz should be 10 Hz, and so on. In Figure 7, the shock-mount resonant frequency should be 9 Hz, not 90 Hz.

The last sentence in the article should read, "For minimum self-noise: Choose high-sensitivity dynamic *or* condenser microphones designed for low-noise performance."

Bruce A. Bartlett
Senior Development Engineer
Electroacoustical Development

db replies:

Sorry 'bout that! Your graphs ran afoul of our art department. And, by leaving out the word "or," we "invented" a new kind of microphone.

MORE ON STEREO MIKING

To The Editor:

In Thomas Ammons' letter (**db** December 1980), he asked what would be the stereo (X-Y) microphone patterns resulting from using M-S technique with the M set in omni or figure-8. Since neither your reply nor the article to which you referred him actually answered the question, I'm enclosing a diagram which shows the gamut of M-S patterns, with the equivalent X-Y patterns to the left.

It might be noted that using a figure-8 for the M, as well as the S (which results in crossed figure-8s in X-Y), was preferred by Blumlein, the inventor of M-S technique. I've found this useful in

"The closest damn thing."

We were recently lucky enough to catch Doug Kershaw on tour. After his show at Harrah's at Lake Tahoe he talked to us about his new Bose® sound system, which consists of four Bose 802 speakers, a Bose PM-2 Powermixer, and a Bose 1800 power amplifier. He's been using the system to amplify his electric fiddle, squeeze box, electric guitar, and vocals.

Q: Doug, you've been playing for a long time. I'll bet you've tried a lot of different kinds of sound gear, haven't you?

Kershaw: Yes, I've used lots of different things and I've spent a lot of time developing my sound. Even then, I could never quite get what I was looking for. But my new Bose system is the closest thing to what I want. The closest damn thing.

Q: What differences have you noticed since you started using the Bose system?

Kershaw: For one thing, it doesn't hurt my ears. You know, I've used some big speakers that have almost busted my ears. I've even put my foot through a few of them. But this is a true sound. It sounds just like my fiddle, no matter how loud I turn it up.

Q: Have you found that you have changed your playing in any way because of how the 802s perform for you?

Kershaw: The attack is easier. It's just easier.

If a Bose system makes it easier for Doug Kershaw, you might find that it can do the same for you. Visit a Bose Pro Products dealer soon and try one out.

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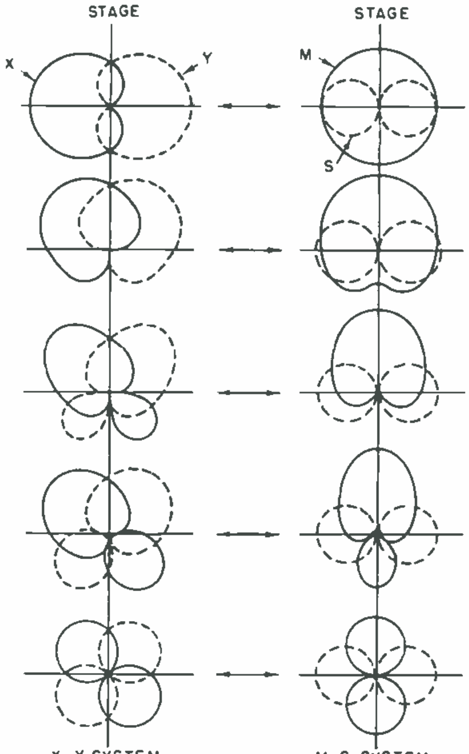
orchestral recordings when one wants to reject some of the early reflections from the sides of the stage. At the other end of the scale, an omni M (or cardioids back-to-back in X-Y) offers an unusual stereo pick-up without the typical loss of off-axis sounds in mono.

Thanks for this opportunity to reply to Mr. Ammons.

LES STUCK
Hyde Street Studios
San Francisco



EQUIVALENT DIRECTIONAL PATTERNS FOR STEREO MICROPHONES

db replies:

And thanks for the additional information. Actually, our application note cited one of the classic M-S combinations. As reader Stuck correctly points out, there are many variations on the M-S theme, and each will produce a different pair of resultant X-Y patterns (and vice versa). In the accompanying diagram, note that the third example produces X-Y patterns that closely resemble super-cardioids. On the other hand, two X-Y cardioids (example 2) create an M-S combination in which the M pattern is cardioid-like, but only down about 10 dB at 180 degrees. The combinations are endless, and given the high reader interest in stereo miking, we'll return to this subject in our next microphone issue (June, 1981).

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Introducing Lifesaver with Disc-ProTec™ Formula. A one-step record preservative, anti-stat, and dry lubricant. One treatment lasts 50 plays or more (Len Feldman* proved Lifesaver lasts 100 plays, but we'll be conservative).

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From the very first audition, a Life-Saver-treated record will have less distortion than even a new untreated record. And 50 or even 100 plays later, it will still outperform that new record, with no build-up of record noise!

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With most Lifesaver treatments, minor scratches and nicks play right through without repeats (no guarantee, but the odds are with us). So if you're into golden oldies, or vintage jazz, or classical music, you don't have to cross your fingers, or do those abrupt fades in the middle of the best part. Nor will you need extra copies of the records on your "heavy playlist."

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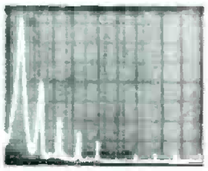
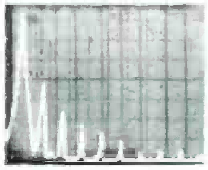
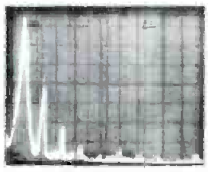
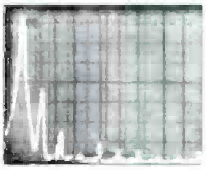
*Here's an excerpt from the Len Feldman report in Audio Magazine. We'll send you the full story with your order.


Harmonic distortion of an untreated disc during first playing.


Harmonic distortion of an untreated disc after 100 playings.


Harmonic distortion of an identical disc, first playing after LIFESAVER treatment. Distortion is immediately reduced.


Harmonic distortion of a LIFESAVER-treated disc after 100 playings. Distortion remains lower than a new, untreated disc.

To The Editor:

It was a real pleasure to read the article "Getting Down to 2 Tracks Fast..." by Ralph Hodges in your December 1980 issue. I didn't think there were any other old "dinosaurs" who would run around a studio with a finger stuck in one ear. Certainly, some of the modern mix-down methods are a marvel of modern technology, but the engineers who grew up on "fix-it in the mix" just don't have the skills to do it any other way.

Those of us who were around in the days of mono and helped to develop the original stereo techniques learned that the most important things to know were the subtle nuances of each microphone in the storage cupboard.

Thanks again for a most enjoyable article.

W. R. Graham
Audio-Visual Supervisor

To The Editor:

How terrific it is to see Fred getting recognition for something only he (and very few others) can do.

Thank you.

David Rubinson
San Francisco, CA

To The Editor:

Three cheers for Fred Catero (Getting Down to Two Tracks Fast and Other Stories, December 1980) via Ralph Hodges. Catero is an old-fashioned engineer, and in this case, being old-fashioned is also being new-fashioned. I heartily agree with his philosophy of recording and wish I had written the article first—although I'm not sure I could have expressed it as well as he did. I'm presently involved with a swing-style

Introducing the Crown PZM,™ the second major advance in microphones in 100 years.

In 1876, Bell invented the first microphone.

Crown now announces the second microphone – the PZM.™

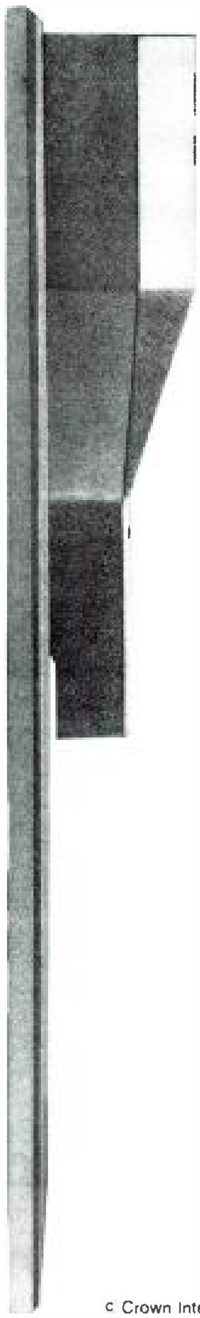
During the last century, microphones have been much improved, but they still employ Bell's basic concept: a movable diaphragm connected to a transducer, the whole assembly intended to be stuck out in the air somewhere near the sound source.

Comb filtering is a side effect of that design that cannot be eliminated. Every Bell-design microphone demonstrates frequency response anomalies because of an inability to satisfactorily combine direct and reflected signals. Phase-induced amplitude cancellation and reinforcement are the inevitable result.

Crown PZM microphones eliminate comb filtering from the primary boundary because they detect sound according to a new principle, the Pressure Recording Process.™ As a sound wave approaches a boundary (wall, table, floor) a pressure field four or five millimeters deep forms at the boundary, within which the direct signal and its reflection from the boundary add coherently and remain in phase.

The Crown PZM™ places a small pressure transducer into the primary boundary pressure zone, eliminating the possibility of phase-induced interference. The PZM concept thus provides a significant improvement in signal quality. Its small profile also improves microphone aesthetics.

The PZM pickup pattern is hemispheric, with no "off-axis" position.



Singers and speakers can move more freely around the PZM. Gain related to distance will change, but not tonal quality.

The PZM responds accurately to SPL up to 150dB. You can put it right inside a drum, a bass fiddle, or a piano.

The PZM hears whispered conversations in an ordinary room at thirty feet. In certain situations where undesired ambient noise can't be eliminated, or in halls with poor acoustics, the PZM probably should not be used – it will pick up everything.

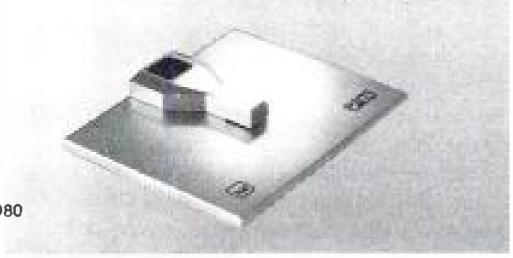
Singers, orchestra conductors, pianists, percussionists, broadcasters have all tried – and praised – the PZM.

Recording engineers find that the PZM suggests new miking techniques. For small groups it now seems that the best place for a PZM is on the floor! Recording and reinforcement may well require fewer PZM mikes.

Several PZM models are now available, including a clip-on and recessed model for permanent installation.

The PZM is changing ideas about how a microphone ought to sound, look and be used. Find out for yourself how it might improve your own recording or reinforcement systems.

Write for information on the PZM. Or call us at 219/294-5571.



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Circle 13 on Reader Service Card

big band whose fourth album I did not record, but happened to be present at the recording session. The band members, who are friends of mine, kept coming to me and asking why everything sounds like "it's being recorded in a box?" It was not my job to be engineer or producer on that session, so I quietly told them I would explain later. After the session I explained that the miking techniques used by the engineer for the session did not permit pickup of the studio ambience. In addition, the studio itself was, to use Catero's words, one of those "semi-anechoic sound-sink(s) that pass for a studio today." The band didn't have a chance. I will be recording their next album—elsewhere.

Also, can you tell us the album number of "Black Pearl" so we can check out Fred Catero's work?

BOB KATZ
Recording Engineer
New York City

db replies:

As of now there is no US distributor for the "Black Pearl" album. As soon as word of one reaches us, we will pass it on to you. If you would like to familiarize yourself with Fred's work, he was the engineer on all of Herbie Hancock's albums, and many of the earlier Santana albums.

db Calendar

APRIL

6-9 **National Noise and Vibration Control Conference and Exhibition.** Hyatt Regency O'Hare, Chicago, IL. For more information contact: NOISEXPO, 27101 East Oviatt Rd., Bay Village, OH 44140. Tel: (216) 835-0101.

12-15 **National Association of Broadcasters.** Convention Center, Las Vegas, Nevada. For more information contact: NAB 1771 N. St. Northwest, Washington, D.C. 20036. Tel: (202) 293-3570.

25 **1981 Midwest Acoustics Conference.** Hermann Hall, Illinois Institute of Technology, Chicago, IL.

MAY

5-7 **National Sound and Communications Conferemce.** Hyatt Regency and Atlanta Hilton, Atlanta, GA. For more information contact: Gerald M. Newman, 222 S. Riverside Plaza, Suite 1606, Chicago, IL 60606. Tel: (312) 648-1140.

12-15 **AES 69th Convention.** Los Angeles Hilton, Los Angeles, CA. For more information contact: Audio Engineering Society, Inc., 60 E. 42nd St., Rm. 2520, New York, NY 10165. Tel: (212) 661-8528.

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.

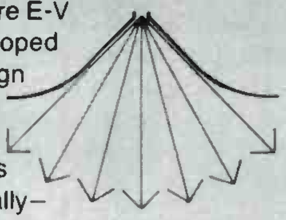
21-7/3 **Audio Workshop.** Concordia College, Moorhead, MN. For more information contact: Audio Workshop, Concordia College, Moorhead, MN 56560. Tel: (218) 299-4201.

The characteristic of a horn that directs all of the frequencies where you want them to go.

Most horns offer some control of the sound pattern they produce. The problem is that frequencies at the center of the pattern are different from those at the edges.

Unless you use **HR Constant Directivity** *horns, that's the problem you'll have.* To the audience this means unintelligible, too bright, too dull, and sometimes just plain bad sound at many seats.

These patented[1] **HR Constant Directivity** horns from Electro-Voice provide full-range frequency coverage and effectiveness of pattern control unheard of before E-V engineers developed this unique design concept.



Demand for the "white horns" has grown dramatically—almost completely by word-of-mouth. Once a sound engineer, musician or facility owner hears the difference **HR Constant Directivity** makes, a new demand is created.

Ask someone who has used or heard them, or buy a pair and try them yourself. You'll probably hear that HR horns are so clearly superior that other choices are obsolete.

Write to Electro-Voice for more information. We'll send you a complete set of Engineering Data Sheets and a paper comparing the today performance of HR constant directivity horns with yesterday's promises. Include $1 with your request, and we will put you on the mailing list for the E-V "PA Bible," a down-to-earth series of papers on the selection and application of professional PA products and concepts!

[1] U.S. Patent Number 4071112

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BARRY BLESSER

db Digital Audio

Distortion

• We have talked about the nature of the noise inherent in the quantization process but we have not mentioned anything about distortion. In one sense, distortion is similar to noise, since both are differences between the input signal and its digital representation. This equivalence does not exist in the analog domain. Analog equipment has different mechanisms for noise and distortion. Noise comes from unwanted signals such as thermal effects, hum, crosstalk, etc. Distortion is the result of a non-linear relationship between input and output.

DIGITAL DISTORTION

We can draw an analogy by asking about non-linearity in the A/D or D/A transfer function. One could have quantization levels that create a gradual non-linearity. However, current specifications of DACs are in terms of ½ LSB deviation between the actual level and the ideal level. The maximum value of the non-linearity, regardless of its shape, is thus limited to this value. So we can conclude that classic harmonic distortion will be on the same order of magnitude as quantization noise. Since these noise values are on the order of 60 to 90 dB, harmonic distortion will be on the order of 0.1 to 0.003 percent, referred to a full-level signal.

With such performance, the issue of harmonic distortion would appear to be irrelevant. Moreover, harmonic distortion can only come from a systematic non-linearity. For example, various-order components of harmonic distortion come from a non-linear transfer curve which contains small quadrature components, as seen in the formula:

$$E_{out} + E_{in} + A(E_{in})^2 + B(E_{in})^3$$

Here, the values of A and B determine the amount of second- and third-harmonic distortion. To see coherent second- and third-harmonic distortion in a digital audio system, the DACs would need to have a similar relationship. In general, there is no such systematic error, since digital errors are either random or concentrated at certain points, such as the major carry. Being random, the errors are described as noise, not distortion.

We now reach an interesting conclusion: non-linearities and noise are essentially the same, except that the former is systematic, and the latter is random. Currently-manufactured DACs, used in both the A/D and D/A process, have an identical specification for these. There is, however, another manufacturing technique which produces DACs with a large amount of systematic non-linearity, and a small amount of random error. These are called segment or piece-wise linear DACs. They are not widely used, and the typical digital audio system is unlikely to have such devices.

LOW-LEVEL DISTORTION

One interesting case of distortion is for low-level signals, as we discussed in a previous column. With a small enough input signal, a sine wave is represented by a square wave. This is because there are only two quantization levels to represent the input. In this case, the quantization error is really a distortion, not a noise. Since this may only happen for signals that are 90 dB below clipping, we usually ignore such distortions. Moreover, the inherent noise in the original analog signal means that such a small signal cannot exist. Hence, more than two quantization levels are spanned.

Most of the interesting forms of distortion in a digital audio system are not from the quantization process. Rather they are from the support process, such as the sample/hold, filters, and other defects. We will explore these next.

HIGH-LEVEL DISTORTION

When the input signal becomes large, there is a tendency for certain stages to begin to show special forms of distortion. We should remember that these distortions will appear to be large when we compare them to the 0.003 percent of the DAC, but they may be small compared to typical audio equipment.

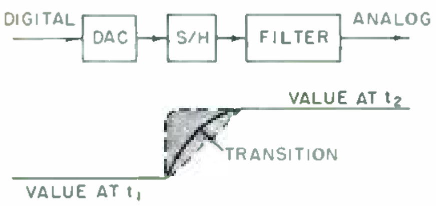


Figure 1. D/A conversion, showing various transitions between t_1 and t_2

Let us consider the sample/hold used at both the input and output. Each must make a transition from the previous value to the next value in a very short amount of time. In a system with a 50 kHz sampling rate, the input sample/hold may only have 5 µsec to make the transition from −10 volts to +10 volts, since the other 15 µsec are used for the conversion. This gives a slew rate of 4 volts µsec. Although the amplifier may be able to move this fast, or even faster, the actual shape of the transition may be slightly non-linear. Instead of achieving the next value, there may be an error. The question of settling time is very complex and we will defer it. Just note that the slew rate must be larger by the ratio of sample time to sample rate. In the above example, the distance that the audio signal would normally go in 20 µsec must be covered in 5 µsec.

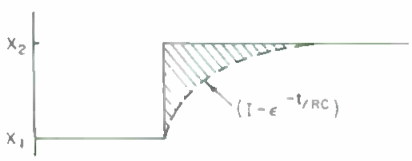
The distortion in the output sample/hold is much more interesting than that of the input. Notice in FIGURE 1 that the sample/hold is *continuously* connected to the output filter. With the input sample/hold, only the final value is used by the A/D. Certain distortions in the process of getting to the final value are meaningless if the final value is actually achieved correctly. With the output structure, not only must the final value be achieved, but the way in which the signal goes to this value is important. The graph in the figure shows an initial value at t_1 and a final value at t_2, with a variety of transition shapes. The full signal is filtered and differences in the transition will be filtered also.

The ideal transition is the square jump in which the signal goes instantly from one value to the next, but no device will actually do this. Another type of transition is an RC exponential; a non-linear transition. We should note that, in terms of what comes out of the filter, a straight line becomes non-linear, and an RC is actually linear. To illustrate this, we will go through the following arguments.

We take the ideal perfect jump as having no distortion and we assume that this produces the reference signal which we will use to compare other cases. Next, we compare this to the RC approach as shown in FIGURE 2. Since the filter is a low-pass, we may argue that the area of the difference is relevant; not the actual shape. A low-pass filter is an averager. The shaded area can be shown to be proportional to (X_2-X_1). Notice that all of the above operations: delay, subtraction, and scaling, are all linear operations. The subtraction is like a difference or differentiation, which is a filtering operation. The net effect can be shown to be a slight error in the frequency response, not a distortion.

FIGURE 3 shows the case with a linear transition. Notice that the area of the shaded region is proportional to $(X_2-X_1)^2$ since both the height and width of the triangle are proportional to the difference in signal values. This figure shows that changes in the difference between X_2 and X_1 produce large changes in the

Figure 2. The ideal transition, compared to an RC exponential curve.



area. Thus, this effect is non-linear and it will produce distortion. The effect is third harmonic, since the shaded area will either add or subtract, depending on the sign of the difference.

We should comment that this error is largest for large differences between neighboring values. Hence, high-frequency signals produce the most noticeable slew distortion as seen from the output. A 20 Hz signal does not change value significantly in a 20 μsec sampling interval, but a 20 kHz signal goes almost from full-positive to full-negative.

In the process of describing the effect, we have used exaggerated figures. Very small regions of slew-like distortion can produce degradation which is larger than that produced by the DACs quantization. The only forgiving aspect of this mechanism is that it shows up with the least likely signals: high level and high frequency. Most high quality digital audio systems will generally show some of this effect, but one requires a good spectrum analyzer to see it.

If you are keeping your wits about you, you may be asking: If the effect produces third harmonic distortion, and if we use high frequencies, shouldn't the harmonic component be filtered out by the filter? A test frequency of 18 kHz in a 50 kHz sampling system will produce a component at 3 x 18 kHz or 54 kHz. This is clearly outside the passband of the output filter. The answer is difficult to demonstrate, but the third harmonic will beat with the sampling frequency to produce a 4 kHz distortion since 54 is 4 kHz above 50 kHz. A 17 kHz sinewave will produce a 1 kHz distortion. The slew distortion is not harmonic but it becomes a low-frequency component. This would not be true for an 8 kHz input, since the third harmonic is 24 kHz which would be filtered out. Also, a 4 kHz input would produce a 12 kHz third harmonic, which would not be filtered nor would it beat with the sampling rate.

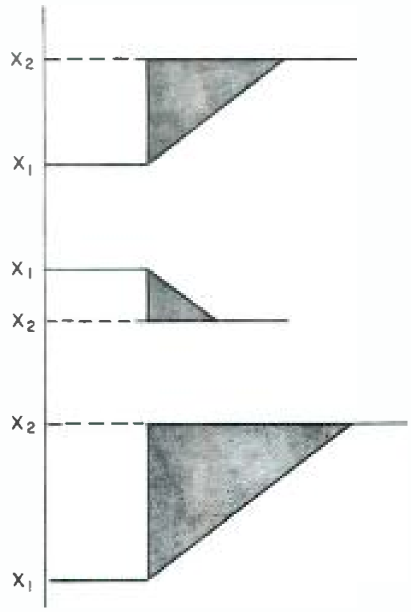


Figure 3. With a linear transition, the shaded area depends on the values of X_1 and X_2

In terms of perception, 16.5 kHz would be very bad since it produces a 500 Hz distortion. The input signal is not audible but the distortion is clearly heard. No signal masking will cover the 500 Hz.

TESTING

It should now be clear that the most difficult test for a digital audio system is a full-level signal at approximately one-third the sampling frequency. If the system performs well in this case, there is probably no distortion problem in the entire chain. It is a simple test and it does not require fancy equipment if you use your ears. Normally, one could not use an audible test since the distortion component was a multiple of the fundamental. The ear, as well as all the support monitoring equipment, generates more third harmonic than the equipment. However, in this special case, the distortion is moved to a new frequency which is very audible. None of the other equipment in the studio can move a single harmonic to a new frequency. Intermodulation tests are similar but they use two input signals to create the beats. In this case, the sampling frequency replaces one of the reference signals. ■

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NORMAN H. CROWHURST

db Theory & Practice

Intelligent Solution of Problems

• The principle we want to explore here, involving both theory and practice in problem solving, can appear in a variety of contexts. Our first awareness of it came in an engineering context, when designing equipment. You would start with theory and run through a series of designs, based on prior data, until you arrived at one that, in theory, was close to the requirements you had established from the parameters of the problem.

Then you went down to the model shop, and got the design made up for testing. More likely than not, the finished product would differ from its design spec, and you would have to adjust the design figures and try again. Eventually you'd get a sample that met the specs, and you'd put it in the system for which it was designed, only to find it didn't work for some reason you had failed to anticipate.

Solving a problem in that way can become a very protracted proposition. What bothered me, as a young engineer, was its indirectness; it was like shooting in the dark. We wanted more direct methods.

Back in my days as Chief Engineer at Tannoy, we developed those more-direct methods and, as a result, we were ahead of our competition for several years simply because we were solving problems while our competitors were trying to write down what the problem was. A reputation for that kind of capability gets around.

Often, the direct method involves different approaches to the same basic problem. Back in those days, the only kind of signal generator was one that used frequency-selective networks with variable feedback. There were variations within that theme, such as beat-frequency generators, in which two ultrasonic frequencies were beat together to get the audible one we wanted, or phase-shift networks that directly selected the wanted frequency. But they had one common weakness. All the reactive networks carefully selected the frequency at which the generator ought to oscillate. Then, the feedback determined whether an oscillation started and built up to a stable amplitude, or whether it died away and quit oscillating. While some excellent generators were made that way, they were always difficult to adjust, and could easily go wrong.

Eventually, someone concluded that we were going the wrong way—or not the best—way about it, and the function generator was born. This uses a whole new approach, which is amazingly more direct. We've covered that before, so we'll not belabor it here.

All of this connects into music synthesizer technology, which didn't get its start until many years later, but which applies the same principle, and carries it further. In those days, the nearest thing to a synthesizer was an electronic organ. And organs of those days used individual generators of the same type; either one for every note on the organ, or one for each of the notes in the top octave, with

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Circle 14 on Reader Service Card

dividers for the lower octaves.

But digital technology, which is related to function-generator technology and is also far more direct in basic concept (if more complicated in the details of its implementation), found a way past that. In those days, if you wanted to tune an electronic organ, you had to adjust the frequency of every note in the first type, or of every note in the top octave in the second.

Digital techology made it possible to use a single, ultra-sonic master generator, and to derive all the notes used from that by division. That was a more sophisticated way of applying the divider principle that had formerly derived lower octaves from the top one, by a succession of divisions-by-2. But now division by any preselected set of numbers was possible by use of a digital programming technique, so that one master frequency could generate all the notes on the musical scale.

If the master frequency was shifted, every note on the instrument shifted to correspond. Thus a whole organ, or synthesizer, can have its pitch shifted with one simple adjustment. Again, a more direct approach to doing the job.

Back to engineering in the lab. You have a piece of experimental equipment in front of you, made to your design, but it doesn't work in the way you predicted. What do you do? Basically, there are two approaches. One is to check a lot of things to find out what is not happening that you predicted should, then apply theory to see why it might not be happening, "zero in" on the problem, and finally make changes to overcome it.

To many, that seems like a long process. They want to get a quicker answer. So they start "trying things." They remember other instances where strapping terminals together, making additional connections, or changing values, solved such a problem. So they start making such changes, willy nilly. If they could just "hit on the right one," it would be quicker than the process we first described.

There are two things wrong with that. First is the fact that usually it becomes a case of "lazy people taking the most pains"; what seemed quicker turns out to take much longer in practice than in theory, and probably never produces a solution. Secondly, even supposing you find something that "works" this way, you don't know why it does. Usually it turns out that you've solved one problem, and made two more that may not show up until later, perhaps when a few million of the products are out in the field!

Having looked at the general nature of such problems and the approach to their solution in an objective or engineering context, let us turn to a subjective context, in which our engineering technology may find a way to help. What we are really talking about is a more-intelligent relationship between man and machine. Too often, we try to reverse their roles. People think creatively, machines don't. But often people who can't (or don't want to) think, expect machines to do the thinking for them.

Many readers know Don and Carolyn Davis, who put on the Syn-Aud-Con seminars. I remember talking to Don one day, quite a long time ago. He had just acquired a Hewlett-Packard calculator for something like $6,000.

Don confessed that in high school and college he was a real dummy in math. It just didn't make sense to him; but since acquiring his calculator, he had sat at his desk "doodling" with it, and came to the conclusion that he was not really a math-dummy after all! He just hadn't been taught right. So, he re-discovered math all by himself, sitting at the calculator keyboard.

He was so taken with the value of this approach (who wouldn't be?) that he wanted to pass it on. He went to schools and tried to persuade them to invest in such an instrument, with little or no success. The problem? Teachers are afraid to venture into unchartered waters, and are not about to turn children loose on such a thing—as Don had done for himself—because to do so would mean a loss of "control."

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Today, I encounter a different problem. For about $9.95, any youngster can acquire a calculator with the four basic functions on it. That's a far cry from Don's $6,000 investment, and so it has become impossible to keep them out of the classroom. Most teachers have given up. Now, it's the parents who ask about the practice of allowing kids to use calculators. They feel that—unless the kids have to memorize the multiplication tables and other things that were required years ago—they're just not going to "learn."

I can't quite agree, and I'm sure Don Davis wouldn't either. If the calculator could really be used as a "cheater," then I'd have to agree. But just as the much-more-expensive one in which Don had invested was a useful tool in learning mathematics, so too can these cheap pocket ones be. It's not *whether* they are used, so much as *how* they are used.

Back in the days when the Federal Communications Commission had just approved the stereo multiplex system that is in common use today, a publisher asked me to get a book ready that would scoop the market on this subject. There were already about three ways to design a multiplex decoder, and the technical publications were full of ads from all the leading tuner-makers. "Ours uses the best system" campaigns gave the impression that any day now, you'd be able to go to your corner radio store and buy one.

Thinking this was so, I wrote to all the manufacturers for information to include in my book, but got no answers. After a follow-up letter, I started getting phone calls, from Los Angeles, Chicago, Philadelphia, Boston—all over. They all said the same thing: "Hop a plane, we'll pick up the tab."

It struck me as a rather expensive way to distribute information; why not just stick it in the mail? I soon found out. Arriving at my various destinations, I was met at the airport, treated royally, wined-and-dined by the executives, and then ushered into the lab, where one or more engineers were trying to make the prototype work. They'd read all the theory, decided on a design approach, but just couldn't make it work.

In each case, I'd spend a few hours in the lab, examine the problem, perhaps help remove a roadblock or two, and leave. I began to realize that the roles of man and machine were being reversed by our society.

In school, students are taught to solve problems according to specific type. The teacher must know how, so he or she can show the student how. Then, the student knows how. But anything that has not been shown, cannot be solved. This is a serious malady in today's world, and total reliance on computers just aggravates it. But the same change that remedied things back then, still can.

People think that anything a computer spits out is infallible, because a computer "can't make mistakes." Maybe some are a bit disillusioned about that, particularly people who spend $100 on their credit card, and get billed for $100,000. That's apt to strain anyone's faith in infallibility. But they still don't ask, "Why do mistakes happen?"

The musician who plays a classic (non-electronic) instrument requires skill and training to master his instrument, doesn't he? Does a person who plays a synthesizer, or an electronic organ, not need such preparation? Perhaps this is a silly question, but I think you get my point. It may be a more sophisticated instrument than its classic forerunners, but still it can only do what the musician makes it do. (Maybe I'll get some arguments on that.)

Mistakes in using calculators or computers happen mainly because the operator hits the wrong button, or hits the buttons in the wrong sequence. Sometimes this happens because the operator doesn't really understand the machine (or maybe even the problem) and is just "trying things."

On an electronic music instrument, the musician/operator can waste a lot of time "trying things." Or, he can spend some time learning to understand the system's sophistication. ■

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MARTIN DICKSTEIN

db Sound With Images

• Most of us think of our eyes as being our most important sense organs. It's true that we learn mostly by what we see, but it has also been found that loss of hearing has a more disastrous effect on the psychological development of humans than blindness. For all you audio people out there, this should be easy to "see." All your critical hi-fi listening might go in one ear and not the other.

It's true that in the early development of the human body, the ear was originally meant to be used for balance. That's why a damage or defect in the inner ear causes vertigo, car-, plane-, and boat-sickness. (Those wild rides at the amusement parks are made to stimulate this sense, and can either titillate or nauseate, depending on your inner ear.) But the ear also developed to hear, and although parts of it are not as useful to the human as they might be to animals, down deep inside (about an inch or so below the outer ear) beats a very sensitive mechanism, which, through all its complex connections, allows the brain to receive those sensations we call sound. Names like ear drum, tympanic membrane, tympanic chamber, cochlea, ossicles, auditory meatus, eustachian tube, oval window, and round window identify a few of the parts that make up the mechanism that allows us to hear.

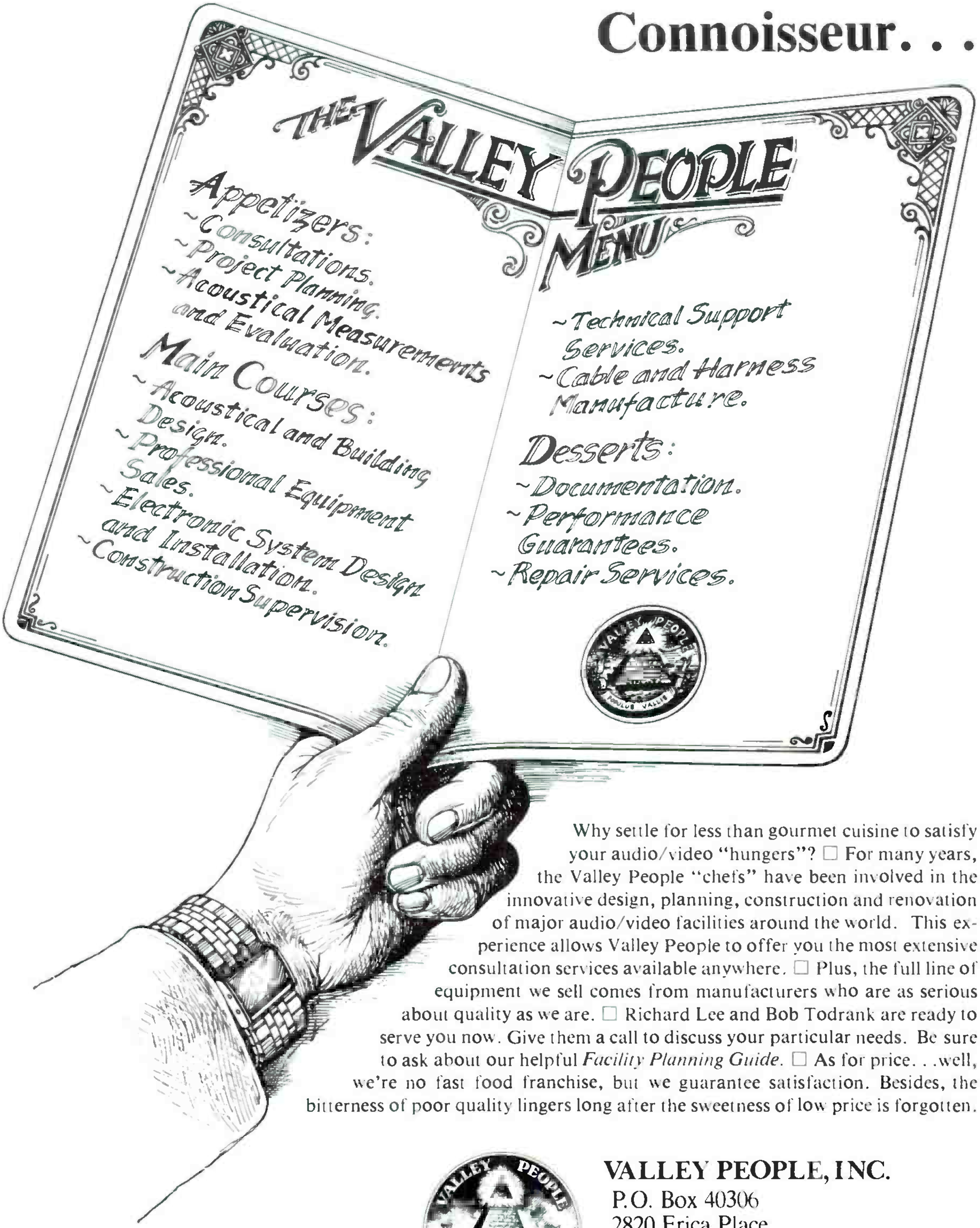
Although we don't realize it (or take it for granted), the entire mechanism is really a marvelous device. While the part of the ear outside the head doesn't work well for us (dogs, for example, can literally perk up their ears to help gather the sound they want to hear while humans have to cup their hands and pull the outer ear away from the head to do the same thing), we can concentrate on one sound even though it may be buried among many others that are higher in level. You might be interested in knowing that the small air space leading from the outer ear to the ear drum has a resonant frequency of about 4,000 Hz. It might also be interesting to note that the ear drum is extremely efficient in that it reflects very little of the incident sound and transmits almost all of it to the area and parts behind it. That part of the hearing mechanism acts like a preamp. Pressure on the ear drum is only about one fifth the pressure passed on to the inner connections which eventually feed through the eighth cranial nerve (there are 12 of these nerves in total) to the lower brain.

With this kind of mechanical advantage, it's quite obvious that the loudest sounds (mostly man-made) could cause damage to the hearing mechanism, so the ear was equipped with two striated muscles (the smallest in the entire body) to try to increase the system's internal friction and prevent the 12,000 little hairs (that's all we have) in the system from being bent out of shape. Sometimes though, the incoming wave front is so strong and so sudden, that the muscles, which need about 60-150 milliseconds to contract, don't react quickly enough and there is some damage.

Consideration of some other small details will further illustrate how marvelous the ear is. For example, the area of the ear drum is about 70 sq. mm. The oval window, through which vibrations are finally transmitted, is only about 3 sq. mm in area. In spite of these tiny dimensions, the approximate minimal energy level of a 1,000 Hz signal that we can sense is 0.0002 dyn/sq. cm. If our hearing were more sensitive, it is conceivable that we would be disturbed continuously by molecular bombardment which might allow us to hear temperature differences. (I wonder if that's how crickets and their kin can tell whether to rub their back legs faster or slower.)

It's also amazing that the human ear can distinguish about 250 distinct intensity levels, and about 1,000 pitch discriminations in the range from 20-16,000 Hz. (Some of us humans have what is known as perfect pitch and many are greatly stressed by screeching cars and subway trains, while others must cover their ears during the sound of rumbling trains or roaring planes.) The human ear can be sensitive to "volume" or "mass" of sound as distinguished from loudness (which phenomena are not easy to explain but for which we now have different meters and different nomenclature).

For the Audio/Video Facility Connoisseur. . .



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The ear has many more marvelous things about it which would take a good deal of time to illustrate, but the reason for all this is to show how much there is that can be different from one ear to the next. Some people need louder sound in order to get the full range they are listening for, while others are so sensitive that they turn high fidelity controls down for comfortable listening. Some of us raise the bass while others raise the highs during the playing of the same piece of music. Some like the deep richness of a "natural" bass tone, while others can't tell if the bass is real or artificially flavored. Some buy certain hi-fi components for whatever quality they are seeking, while others will think lo-fi systems in a single housing totally adequate. It's all a matter of individual taste, and that's why there are so many manufacturers and models of speakers and components available in the market place. There are also some individuals who think that price alone makes the difference, and will buy only the most expensive items for the home system even though they may not be able to distinguish between a violin note and a goose honk.

This leads to the next point. In recent columns in this corner, the question of sound during audio-visual presentations was raised. It seems only fair that the time, effort, and cost of the visual part of the show be matched by the audio part. It seems such a shame and waste to have so much put into the pictures and so little into the sound. A great visual presentation can be badly hurt by poor sound, and a fair visual show can be killed by bad sound. But what is good sound?

Shortly after several of these columns came out,inquiry was made by readers as to what speakers we would recommend for audio-visual shows. We are very reluctant to suggest any particular brand of speaker, or model. Not all the speakers available are acceptable, of course, but to suggest certain units in preference to others would be unrealistic and unfair. What we would like to do, though, is to suggest some of the parameters that should be used as guides for selection of speakers to go with shows.

For instance, think of the size of the room in which the show will take place. Each situation might be totally different from the next. If the room is very large, and the audience is expected to fill the room, more sound has to be distributed than if the room is small and will contain only a few people. It's always best, if possible, to have the loudspeakers at the front of the room near the screen for greater realism. Sometimes this is not possible, because a lectern with a live microphone is right in line with the loudspeaker and might result in feedback. Perhaps the speakers can be moved off to one side, farther away from the screen and in line with the lectern to minimize the possibility of feedback. This might result in sharper angling of the speakers toward the audience. Be sure however that there is not a "hole" in the center near the front. If the stage platform is raised fairly high, perhaps the speakers can go under the screen. Just remember that there should not be too much sound for the first row in order to get a little sound into the last one. Perhaps the speakers can be raised above the first row and then angled to give enough sound for all. It all depends on the room and the setup.

Speakers should never be underpowered. If the room is large, a few simple calculations considering the distance to the last row and the level desired will give the requirement the speaker should meet to provide that sound. Like with all other sound equipment, never work a speaker at its top edge of operation. If you raise the volume, the speaker should not break up or crackle. Also, keep in mind the amplifier that is feeding the speaker. Don't run that unit at its top, either. Consideration must also be given to hard surfaces that can reflect sound. For example, putting speakers in a corner of two hard walls might possibly provide more bass than you want or need. Carpeting and drapes will absorb some of the highs and leave you with more boom than you want.

Speakers should also be efficient so that the amplifiers don't have to work so hard to get the volume out. If you find the speakers can handle them and need more power, use more powerful amps. There are many criteria to finding the right speakers, but they should all be full range, especially if you have a good music score to play. You can get away with smaller range if all you have is a voice track, but there are a lot of people out there who consider themselves audio or hi-fi experts because they have "good" systems at home, and they are all in your audience. Don't let them down. Provide the best sound possible, to all of them. One last thing to consider. Don't check your sound level in the back of the room without thinking of the people in the audience. This does not mean you have to wait for them to get seated before you set your levels. What it does mean is that they will absorb a great deal of sound when they are in their seats, and you should set your level during testing accordingly higher than with the house empty. You might think of having a member of your crew in the back with a walkie-talkie when the show starts and again during the presentation to keep tabs on level and let you know whether to raise, lower, or leave it alone. He (or she) can also check on the slides and films at the same time, so that you know you're doing your best to give the attendees the best audio to go along with the best visual presentation. ■

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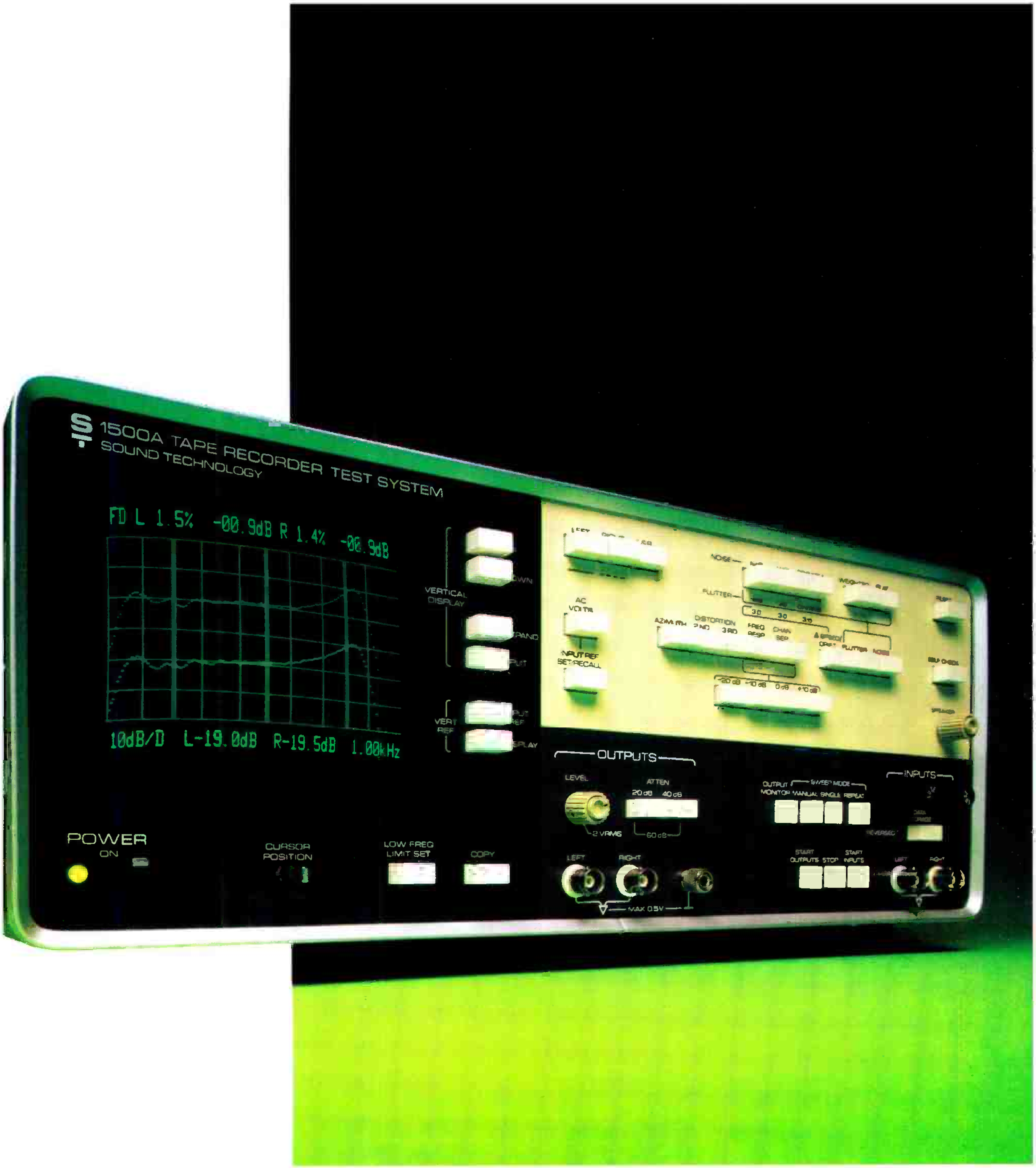

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It's the first microprocessor controlled audio measurement test system. It can do in minutes what used to take hours with more conventional and ordinary test set-ups. And, it can show you things you've never seen before.

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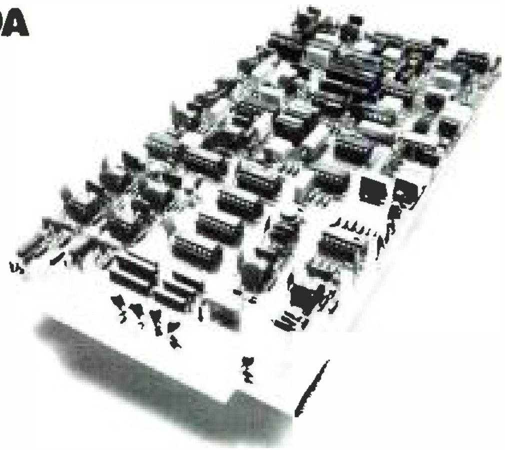

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Conceived to be the ultimate precision test instrument for tape recorder analysis, the 1500A evolved into a comprehensive audio test system for many applications. Here's just a small sample of the varied jobs it will do:

- Complete tape recorder mechanical and electronic performance checks
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db New Products & Services

CASSETTE COPIER

• The Telex Copyette 1&3 is an economical high-speed cassette copier capable of copying three one hour cassettes in less than two minutes in a single pass. No special training is required to operate the Copyette 1&3. The operator simply inserts the original cassette and from one to three blank cassettes, then presses a single button. All cassettes automatically rewind to assure that all cassettes are copied from the beginning. Copies are then made of both audio tracks of the original on a single pass. When copying is completed, all cassettes again automatically rewind to the beginning.
Mfr: Telex Communications, Inc.
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GRAPHIC EQUALIZER

• The GE-3 graphic equalizer, like its predecessor the GE-2, has a combination of octave and one-half octave filter bands. The GE-3 provides the equalization capabilities of a one-half octave filter set at frequencies below 250 Hz. Above 250 Hz, bands are set on octave centers. This combination of octave and one-half octave equalization provides high resolution in the low frequencies where EQ problems are likely to occur. On the rear panel are the input and output connectors that allow the GE-3 to interface with a variety of sound systems. The GE-3 provides both balanced and unbalanced inputs as well as outputs. The balanced connections are made via standard XLR type jacks, while the unbalanced are made via ¼-inch phone jacks. Both the unbalanced and balanced outputs are capable of driving load impedances of 600 ohms or greater to full rated output levels without distortion, and all inputs and outputs are protected against potentially damaging levels or loads.
Mfr: Cerwin-Vega
Price: $550.00
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WANTED

A magnetic recorder with a frequency response of at least 100 Hz to 10 KHz (-4db), with a signal-to-noise ratio of 20 db at 1 KHz. DC record bias and erase. Head life must exceed 5 hours per pole piece. Wow and flutter should be lower than 10%, with a tape speed of not more than 3 feet per second and a record time of at least 30 minutes.

Sound ludicrous? If **db** had published 50 years ago, the beginning of the steel tape era, such a want-ad would have also appeared farfetched, because of its outrageously high specs!

We've come a long way since the "Blattnerphone" steel band recorder of 1932: the 1936 German Magnetophon audio tape recorder (the world's first modern hi-fi magnetic recorder), the American wire recorders of Marvin Camras' Armour Research Institute, Harold Lindsay's 1947 Ampex Model 200A audio tape recorder, 3M's outstanding developments in both audio and video tape since 1946, to name just a few of our industry pioneers.

The Museum and Archives of Magnetic Recording, an industry-supported project to preserve our recording heritage, wants to hear from you about the whereabouts of old wire and tape recorders (audio and video), old tape recordings, photos, and documents.

Please contact
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MUSEUM AND ARCHIVES OF MAGNETIC RECORDING
c/o Ampex Corporation
401 Broadway, Redwood City, CA 94063
telephone (415) 367-3127

COMPRESSION DRIVERS

• Designated as EAS-45HD05EX and EAS-60HD05EX, these new drivers are designed to improve reliability and acoustical performance in professional speaker systems. The compression drivers will withstand high voice-coil heating while exhibiting an improvement in output power linearity. Among the features of the drivers: high power handling capability (with ferrofluid); titanium diaphragm for long life; rhombic edge; multi-slot phasing slug, and a high efficiency magnet structure.
Panasonic Company
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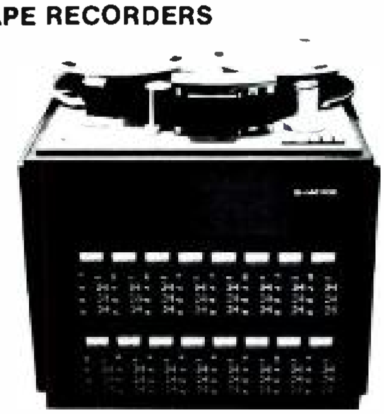
MULTITRACK TAPE RECORDERS



• The SCM 381 multitrack tape recorders are available in 8 track and 16 track one-inch formats, and in 16 track and 24 track two-inch formats. These machines feature very low wow and flutter, low noise, and small size. Full remote control, varispeed and search to zero facilities are standard, as are separate sync outputs on each track.
Mfr: Soundcraft Electronics
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db Editorial

NEXT MONTH, the topic at the fourteenth annual Midwest Acoustics Conference will be "Electronic Modification of Musical Sounds." According to a recent press release, conference topics will include: characteristics of musical sounds, history of electronic sound modification, electronic sound modification for the performing musician, time domain and frequency domain sound modification, nonlinear modification of sound, synthesizers, and a look at future trends. For time-and-place details, see the MAC ad on page 10.

And for still more on the subject, see this issue of **db**, where we begin our two-part series on "Music in the Studio: Electronics & Acoustic."

Electronic music has been around almost as long as electronics, though many of us in the recording studio think of it as something invented by Dr. Robert Moog and Walter (now Wendy) Carlos. The Carlos "Switched-on Bach" album brought electronic music out of the laboratory and into the studio, where at first it created a sensation. But it soon faded into a semi-retirement of sorts, as listeners grew bored with the "silly piano" antics of the legion of Carlos imitators.

Within recent years, electronic music has come back again, and this time it is apparently here to stay. The novelty aspect seems to have worn off, and the electronic music synthesizer is now taken seriously as a creative tool on stage and in the studio.

The synthesizer is now a likely candidate for a piece of next year's studio equipment budget. Its versatility makes it an attractive client pleaser—especially if there's someone around who understands how to make the damn thing work!

The vocoder is perhaps the prime example of a synthesizer that needs to be understood. Given the right inputs, it can do wonders; with the wrong inputs—nothing! Since one input will be used to modify the sounds of another input, it follows that the two input signals must bear strong spectral and temporal relationships, but that's another story. Specifically, its Felix Visser's story. This month, author Visser gives us a detailed look at the vocoder's insides, and helps unravel some of its mystery.

Electronic music brings with it the capability for extending the limits of the frequency spectrum of reproduced music. According to Michael Rettinger, electronic music and digital recording technology require improvements in studio monitoring systems, to meet the demands of a wider-range frequency response. At low frequencies, better response means moving more air; for high-energy output at 25 Hz, as many as eight 18-inch woofers may be required. Taking note of the physical impracticality of this, Rettinger proposes a ceiling-mounted folded horn system that cuts down on the number of speakers required, and enables us to construct such a system within a reasonably-sized control room.

More often than not, the synthesizer is tuned to the traditional well-tempered scale given to us by Bach (Johann Sebastian, not W. Carlos). But why not a metric scale to suit the needs of the metric movement? Why not, indeed? The synthesizer makes it easy enough, as Hal Lion and Jim Fox discovered during a recent recording project for the US Metric Board. Once they figured out what a metric scale should look like, it was easy enough to hear what it would sound like on the synthesizer. Although there's little chance that metric music will sweep the country, it's just one more example of how the synthesizer may be put to work in the studio.

One of the earliest predecessors of today's synthesizer was Thaddeus Cahill's Telharmonium, which consisted of some 200 *tons* of alternators, dynamos, transformers, gear and shafts, plus a two-rank keyboard. Suitably impressed, Ferruccio Busoni wrote, in 1907, that "... music is born free, and to win freedom is its destiny."

In the early '80s, it's difficult to appreciate the "freedom" that comes with a 200-ton instrument. Needless to say, today's synthesizers are a bit lighter, and certainly a lot more sophisticated. The latest generation is of course built around the microprocessor, which weighs a lot less than any collection of gears and shafts around. To find out just what we can expect from these grandchildren of the Telharmonium, Ralph Hodges visited a few west coast manufacturers, to bring us up to date on the latest synthesizer hardware (*and* software).

For still more information on electronic music, just check in with any of the manufacturers in our synthesizer directory also found in this issue of **db**. ■

FELIX VISSER

Vocoders

The secrets of the mysterious vocoder revealed.

IF THERE WILL ever be a list of electronic audio instruments which have caused confusion in the professional audio and music industry, the vocoder will definitely be on it. Recently, the mysterious vocoder has become a desireable, though costly, recording studio instrument. However, the mystery element persists, often because the actual purpose and musical versatility of the vocoder are somewhat—or often, totally—misunderstood.

THE BASIC PRINCIPLE

Though it is hardly possible to list all efforts made in history to synthesize human speech, the name of Homer Dudley is inseparable from today's vocoder. In 1936, he patented his apparatus to analyze and remake speech, which he called a Vocoder, because it was based on the principle of coding the voice and then reconstructing the voice in accordance with that code. (So please, no more "Vocorder"—it has nothing to do with recording the voice!)

The principle of the Dudley vocoder, where speech analysis is performed by a set of parallel-band filters, is commonly called the channel vocoder, as opposed to the formant vocoder, where the resonances produced by the oral and nasal cavities are simulated by several tunable-formant or resonance filters. Being merely a speech synthesizer, instead of an analyzer/synthesizer combination, the formant vocoder is less interesting for music applications and therefore will not be discussed further in this article.

Basically, the channel vocoder can be divided into two main section: the analyzer and the synthesizer, as shown in FIGURE 1. Each consists of identical filters covering a specific frequency range. The analyzer filters divide the speech spectrum into narrow bands, from each of which a voltage is derived. The voltage is proportional to the energy in the band.

In the synthesizer, a second group of filters divides the spectrum into the same narrow bands, and is followed by an amplitude-controlling device, such as a VCA (voltage-controlled amplifier or attenuator).

By connecting all analyzer control-voltage outputs to the respective VCA control-voltage inputs in the synthesizer section, the speech spectrum can be imposed upon a carrier signal (a musical instrument, for instance) which is fed into the paralleled audio inputs of the synthesizer's filter bank. This creates the almost-classic "talking music" effect.

A similar effect can be achieved by the so-called "mouth tube" or "mouth bag." Here, the carrier sound is acoustically injected into the speaker's mouth by a flexible tube, and thus can be more-or-less articulated. Quite apart from unpalatable side effects, such as jaw-muscle spasms, this is a crude way of imposing speech upon another sound.

It should be made clear that the vocoder deals only with harmonic structure, and in no way is concerned with fundamentals, such as the pitch of the voice to be analyzed. Therefore, a circuit which can be very useful when using the vocoder as a voice synthesizer/processor is the pitch-to-voltage converter.

The only pitch-determining component is the fundamental of the carrier sound (the artificial vocal cords). When this pitch is changed, the pitch of the voice will change accordingly. When the pitch of the carrier is kept constant, a change in pitch of the real voice will only result in a different timbre of the synthesized voice. So, in order to generate complete, melodic speech, a device will be needed to change the pitch of the carrier in accordance with the pitch of the real voice. Such a device could be a pitch follower/extractor, as it is generally called. By changing the ratio between real-voice pitch and conversion factor, either linearly or non-linearly, many unrealistic but interesting artificial voice effects can be obtained.

THE ANALYZER

In the Syntovox 221 vocoder, which consists of 20 analyzer and 20 synthesizer channels, each analyzer channel can be split into three subsections: a band filter, a full-wave rectifier, and a low-pass filter with an LED readout. The combination rectifier/low-pass filter is also known as an amplitude demodulator or envelope follower. The analyzer section is shown in FIGURE 2A. The number 20 is arbitrary, and not a scientifically-determined minimum or maximum. (In the case of the Syntovox 221, it was partly inspired by the available standard matrix format chosen to interconnect the analyzer and synthesizer.)

If the vocoder is to be used exclusively to reconstruct intelligible speech, 15 to 16 filters are sufficient to cover the necessary frequency range from about 100 Hz to 3 kHz, with a resolution of 25 percent, which gives us a one-third octave filter spacing. This will provide a fairly-accurate picture of the speech spectrum. However, extending and subdividing the total frequency range beyond the 3 kHz limit will add more definition

Mr. Visser is the president of Synton Electronics B.V., Bruekelen, Holland.

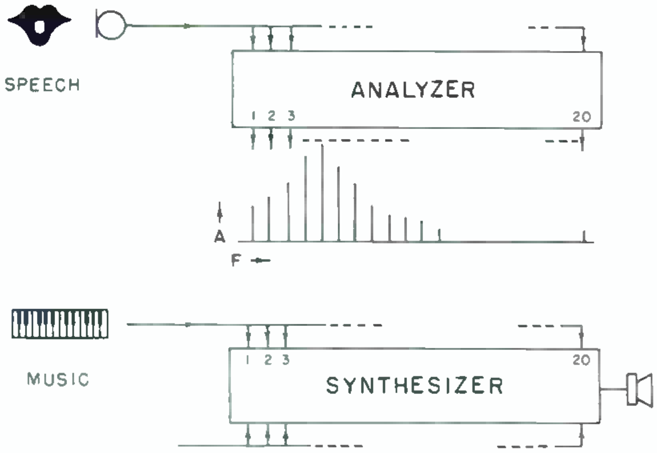


Figure 1. The basic building blocks of the vocoder.

and clarity, specifically where fricatives and sibilants are involved.

An ideal filter would exhibit a flat response within, and infinite attenuation outside, the pass band. In the Syntovox 221, a compromise has been found by giving the filters a relative bandwidth which is narrower than their one-third octave spacing, in order to achieve as little overlap as possible. Moreover, the filters—which are eighth-order—have a very rapid roll-off of 48 dB-per-octave, which increases to 54 dB-per-octave over the first octave.

Although the dips in-between the narrow-band filters will affect the true response of the input signals to both analyzer and synthesizer, they greatly improve intelligibility and effect.

For designers of vocoders and speech synthesizers, it is unfortunate that the human ear is very sensitive to phase relations between spectral components. Frequencies, approximating the resonance frequency of a filter, will be subject to substantial phase shift, causing a different timbral perception, even though amplitude relations within the spectrum have been hardly affected. This phase shift side-effect will be more dramatic the higher the order of the filters, thus creating the unnatural speech effect that is a characteristic of the vocoder.

As with most man-made contraptions, the vocoder is full of compromises. But reducing the filter slopes, and thus reducing phase shift effects, would not result in a better vocoder. Large, overlapping areas create a blurred speech response, along with a poor effect response when the vocoder is used in music.

One more compromise in the analyzer section is the low-pass filter following the full-wave rectifier. In order to obtain rapid response of high frequencies and transients, this filter must be as fast as possible. On the other hand, a large ripple margin will definitely create intermodulation effects, which can be very disturbing.

The problem can be more-or-less resolved by adapting the cut-off frequency of the low-pass smoothing filters to the audio pass-band of the preceding analyzer filters. By choosing a roll-off point about ten times higher than the center frequency of the audio filter, an optimum can be found with respect to the phenomena of transient response and intermodulation as explained previously.

Finally, the LED control voltage read-out array was not only applied for its anticipated appeal—an expectation which has been met—but because it definitely has the function of displaying the spread, and to a certain extent the amplitudes, of the spectral components in the analyzed signal.

To cover the whole audio frequency range with a set of 20 filters, spaced at one-third octave intervals, would not be possible. Since the vocoder is only concerned with the harmonic structure of speech in the first place, it is not necessary to maintain this resolution either, so by designing the lowest and highest filters as low-pass and high-pass, the total audio range can be covered from about 20 Hz to 18 kHz.

THE SYNTHESIZER

A description of the synthesizer filter bank will be easier, now that it is clear that the audio filter section is a replica of the analyzer filter bank. The only practical problem within a certain budget is the fact that both sections should be exactly the same, which means that only a small tolerance in component spread can be allowed. Any deviation from the analyzer center frequencies will cause unwanted formant shift.

In FIGURE 2B, the basic layout of the synthesizer section is shown. FIGURE 3 illustrates the performance of the filter bank in both the analyzer and synthesizer sections. The carrier sound is split up by the synthesizer filter bank and these signals, after being processed by the VCAs, are summed, and become the vocoder output signal. All control voltage inputs of the VCA bank are supplied with an attenuator followed by an LED readout. These attenuators can be used to "equalize" the vocoder effect.

The trickiest part of the synthesizer section is the voltage-controlled gain device following the filter. In order to copy the spectral image in accordance with the control voltage delivered by the analyzer, each synthesizer filter is followed (or preceded, as sometimes is done) by a voltage-controlled gain device allowing spectral components of the artificial vocal cords (the carrier) to pass through to the output of the vocoder. Such a gain cell could be a VCA, an OTA (operational transductance amplifier) or an electronic switch controlled by a pulse-width modulation (PWM) system.

The PWM's amplitude control is achieved by simulating a variable resistor in the audio path, by means of an electronic switch controlled by a high-frequency generator. This switch is alternately opened and closed, either by a narrow pulse whose repetition rate can be voltage controlled, or by a voltage-controlled duty cycle rate.

Figure 2A. The Analyzer section.

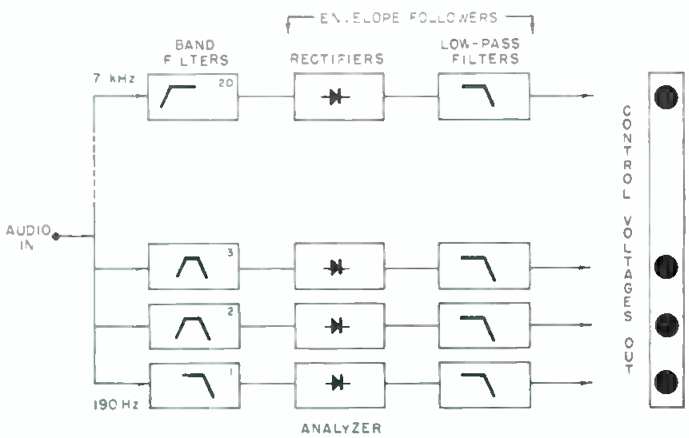
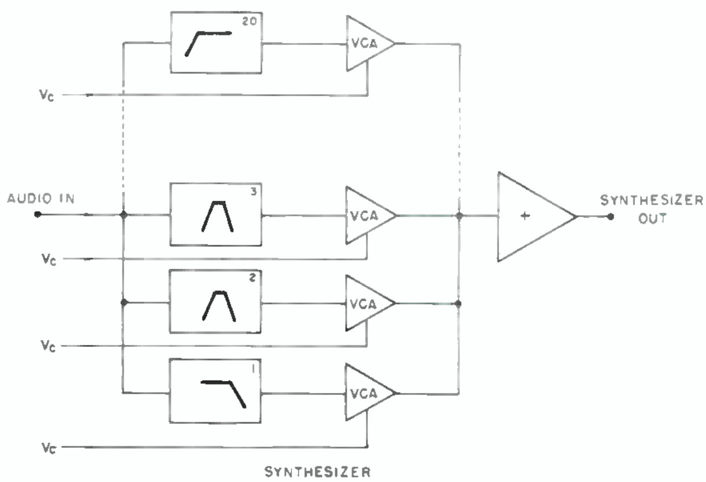


Figure 2B. The Synthesizer section.



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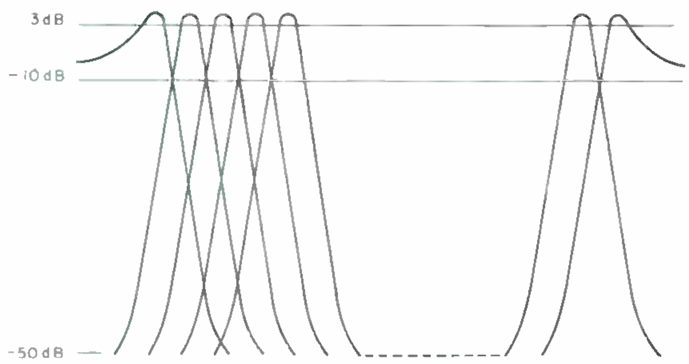


Figure 3. The response of the filters in the Analyzer and Synthesizer sections.

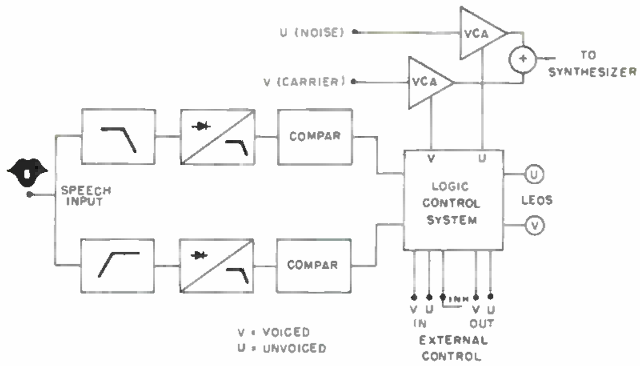


Figure 4. The Voiced/Unvoiced Detector.

VOICED/UNVOICED DETECTOR

The analyzer/synthesizer sections provide a straight-forward vocoder system, with which fairly-intelligible speech and good musical results can be obtained. However, the addition of a voiced/unvoiced detector will add even more intelligibility and clarity to the synthesized vocal effects. FIGURE 4 shows a block diagram of such a detection system, to which many variations and additions can be made. The detector can discriminate voiced and unvoiced sounds by continuously comparing the energy in two different frequency bands. One band is 30 to 800 Hz, and the other is 2 kHz and up. The decision of the detector is based on the assumption that voiced sounds have less high-frequency energy than unvoiced sounds, which fortunately is true for almost all speech sounds. Almost, because composite sounds such as "V" and "Z" can create instability of the detector when pronounced over a long period of time. Under normal speech circumstances, the detector, responding very rapidly to changes of the energy in these two bands, will switch alternately between voiced and unvoiced, and neither the indecision nor the transitions will be audible.

In order to extend versatility in the electronic music studio, and with computer applications in mind, the detector should be equipped with control inputs and outputs, as well as an inhibit input when the vocoder is to be used with other triggering devices.

Another possibility to add intelligibility to the vocoder effect has been provided by Harold Bode, who patented the clever solution of adding the high-frequency end of the real speech spectrum to the vocoder output signal. FIGURE 5 shows the basic setup (U.S. patent 4,158,751).

To provide the necessary high-frequency spectrum for fricatives and sibilants without using an expensive detection system, noise may be added to those synthesizer filter channels that are above 2 or 3 kHz. The noise is constantly present, and it will feed through to the output when high-frequency components in the voiced spectrum open the high-frequency VCAs in the synthesizer. In general, this solution offers better intelligiblity of the synthesized voice effects.

CARRIER SOUNDS—THE ARTIFICIAL VOCAL CORDS

The carrier sound, which will replace the signal normally produced by the vocal cords, is subject to certain conditions which will be discussed later on. For speech synthesis, it is necessary to generate a carrier signal similar to that of the vocal cords, as shown by the functions in FIGURE 6.

These waveshapes each give a different speech result because of their respective spectra. A very narrow pulse signal will provide a synthetic voice of piercing, rattling quality, because of its very strong high-frequency spectrum. A sawtooth and a spaced-sawtooth will give a more pleasant, mellow-sounding voice quality. The pitch generator should be controlled by external sources, such as pitch followers, envelope followers, low-frequency generators and random generators. All these control sources can take away much of the static, machine-like impression that vocoder speech usually makes. It is an interesting experiment to note that a voice will sound less artificial and even more intelligible when the pitch is modulated, instead of being kept constant.

When using external carrier sources, such as musical instruments, an automatic bypass circuit can be of great help to bypass or cross fade the carrier to the output when no analyzer input signal is available. Such a circuit has been incorported in the Syntovox 221, and is called a "fill-in." Other manufacturers supply similar facilities, labelled "silence bridging" or "pause stuffing."

CONTROL VOLTAGE PATCHING

One aspect of the vocoder still to be explained is the interface between the analyzer and the synthesizer. As explained previously, and shown in FIGURE 1, all control voltages generated by the analyzer section have to be fed to the control voltage inputs of the gain cells in the synthesizer section.

Figure 5. The dashed-line path improves inteligibility by adding high-frequency audio components to the vocoder output.

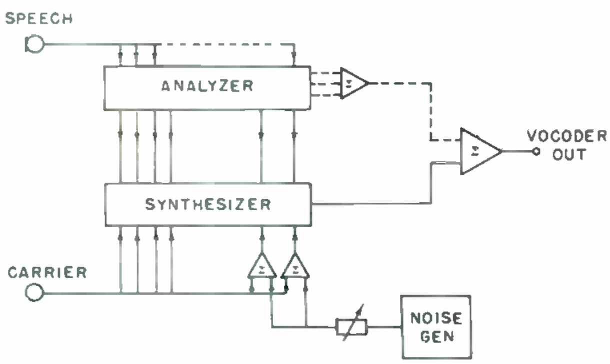
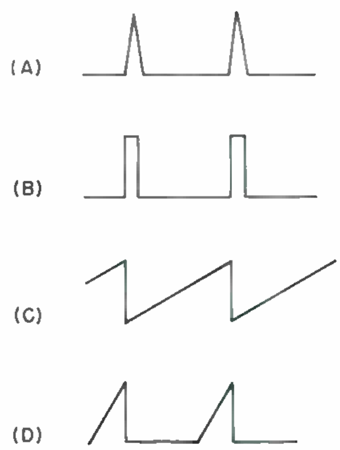


Figure 6. Various carrier wave shapes which may be used to simulate vocal-cord sounds.



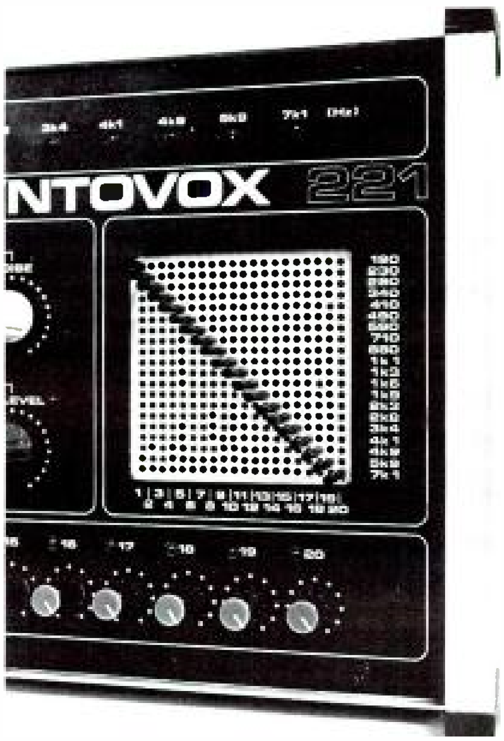


Figure 7. A 20-x-20 matrix connects the analyzer and synthesizer sections of the vocoder.

One of the advantages of not connecting them permanently is that both sections can be used to control, or be controlled by, external equipment. Also, this creates the possibility of connecting the analyzer outputs to other than their respective synthesizer channels. The problem of choosing a way to provide this facility is purely practical, and few manufacturers of larger vocoder systems have gone to this trouble. One way of providing a compact and versatile patching system is the matrix, which allows optimum freedom of routing control voltages. In addition, all outputs and inputs of the analyzer and synthesizer should be available for use in electronic music studio setups.

FIGURE 7 shows the matrix which connects the Syntovox's analyzer and synthesizer sections. In FIGURE 8, all sections discussed so far are shown in a complete block diagram.

One of the most obvious applications of the matrix is formant shifting, which means that formants (typical resonance peaks) are transposed to frequency areas other than where they originate. This feature makes the vocoder an interesting instrument with which to generate different types of voices. Though it is often suggested that the sex of the voice can be changed by shifting formants and raising pitch, this is not true. However, the character of the voice can be changed dramatically, and a certain touch of different age can definitely be achieved.

APPLICATIONS

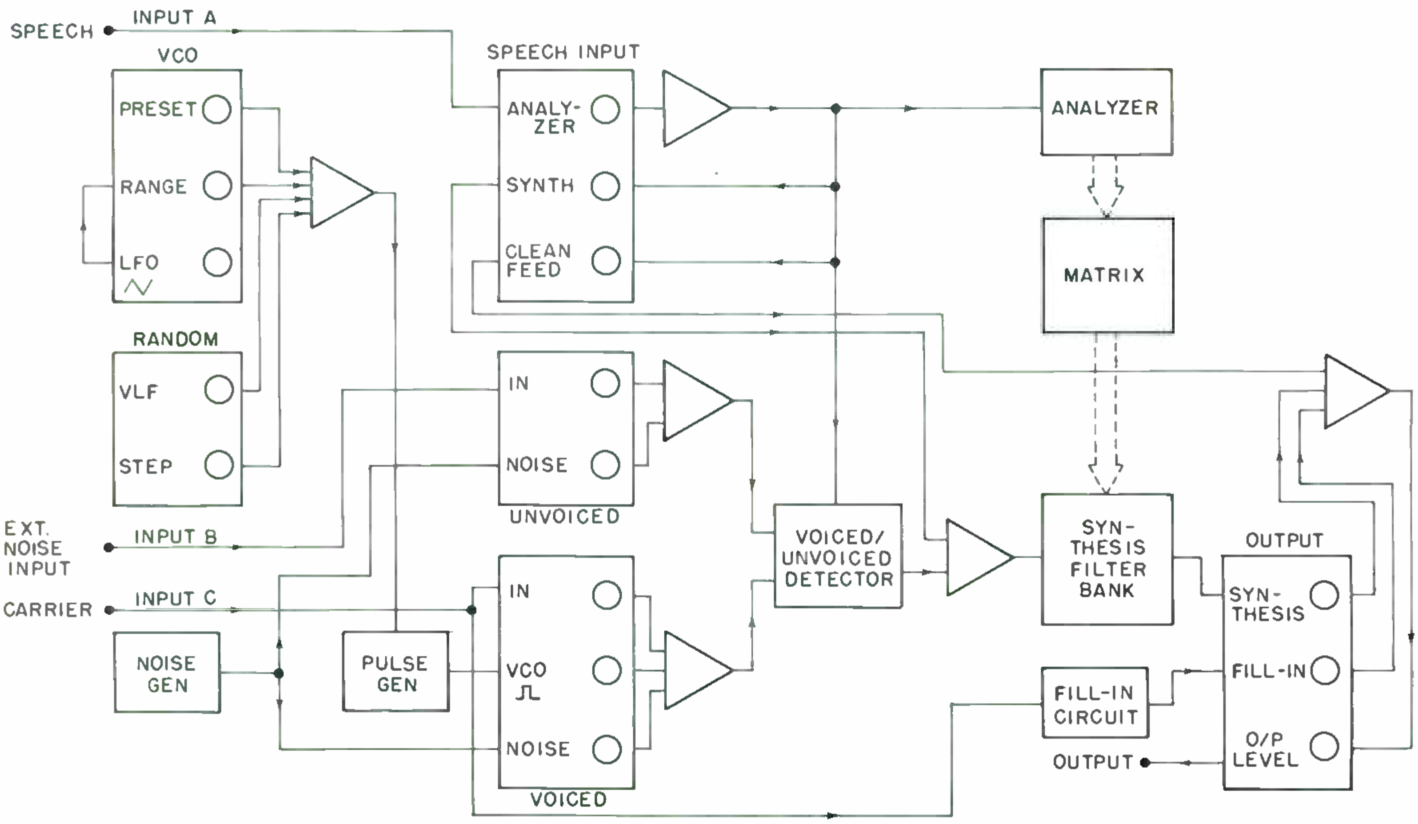
When marketing vocoders, one of the main problems encountered is that it is a typical *two*-input device. The significance of this is not always fully appreciated. In any demonstration, it is almost impossible to show the versatility of the vocoder, since every carrier creates its own effect, and every modulator creates a different effect with the same carrier.

The only thing which can really be made clear to interested people is some basic information on intelligibility, response, and general sound quality.

Japanese manufacturers have appreciated the two-input dilemma, and equipped most of their vocoders with some kind of keyboard instrument, producing organ-like or string-like sounds. They too must have realized that keyboard players belong to the happy few who can play and speak or sing at the same time. However, this conception can easily lead us to the conclusion that a vocoder is a typical keyboard instrument extension, which it isn't. Though it is true that in many cases the voice/keyboard combination gives the most recognizable response within a relatively short setup time, the vocoder makes it possible to use almost *any* sound source to modulate any other sound source.

There are restrictions, especially concerning frequency spectrum and synchronism between sound sources. In order to

Figure 8. A simplified block diagram of the Syntovox 221 Vocoder.



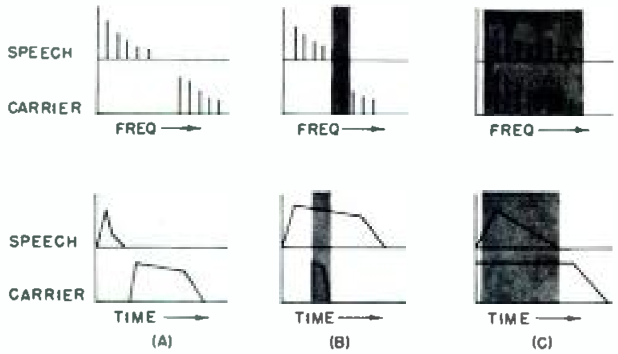


Figure 9. Optimum vocoder performance requires an overlap of spectral and temporal content. In other words, the carrier signal must be harmonically rich, and occur concurrently with the speech input signal. (A) The conditions shown will produce no effect. (B) The two narrow shaded areas indicate a slight effect, which will occur during a brief time interval. (C) Maximum effect is achieved, due to a complete spectral overlap, which occurs continuously.

Figure 10. The Syntovox 221 20-channel electronic effects Vocoder.

get the best effect, it is necessary that both speech and carrier overlap spectrally and in time, which is graphically illustrated in FIGURE 9. A practical situation may make these examples more clear: a poor effect—or no effect at all—will be your reward for trying to modulate the sound of a bass drum with that of a flute, or vice versa. The vocoder user will also get into trouble when trying to impose speech upon a sine wave, or other pure sound.

Problems are bound to come up when modulator and carrier are of short duration and not in perfect sync. A recorded guitar track, producing a sequence of rhythmically limping chords, is not the right carrier to start with, unless the owner of the voice who is going to try to modulate the guitar sounds is alert enough to limp along in sync. Therefore, when a vocoder is used with previously recorded tracks the user may be disappointed with the results. The best way to create interesting vocoder effects is in a live situation, when there is artistic feedback between carrier and modulator. Truly, this is one of the most attractive sides of the vocoder. Being a real-time instrument, it can be used to instantaneously voice-control the timbral quality of sounds produced by electronic music instruments.

Here, the nature of the vocoder touches one of the marketing problems. Due to its complexity, the vocoder is often classified as a piece of high-technology hardware, to be sold to recording studios. On the other hand, the vocoder demands a thorough—almost musical—training of its user, which would make marketing aimed at the music industry seem the most logical way to go.

A sensible way to outline the possibilities of the vocoder may be by placing them in different fields of applications. In the first place, there is the scientific sector, where the vocoder can be a suitable instrument in speech or phonetic research. The effect of transposing formants can be simulated easily by connecting analyzer outputs to different synthesizer control inputs. Vowels can even be inverted by making partial cross patches on the matrix. A very interesting project at the Utrecht State University in Holland tested to what extent speech intonation is important to the intelligibility of the message. This was done by sampling real speech at a 10 kHz rate, and by changing the pitch of the synthesized voice under computer control, which allowed the intonation patterns to be varied every 2 milliseconds. Another purpose for which the vocoder can be used is speech training by producing properly-pronounced vowels and comparing the trainee's pronounciation of them. This can be achieved under computer control by digitizing the analyzer output information and comparing it to digitally-stored information.

A very appropriate application of the vocoder is in the field of animated film and sci-fi productions, making different alien voices by shifting formants and modulating the pitch generator in several ways. Since it is relatively simple to create numerous voice characters with the vocoder, it is strange to note what troubles sound engineers and producers go through, making their movie character voices with the help of ring modulators, filters, envelope followers and shapers, almost desperately trying to avoid the use of the vocoder.

It can be useful to process a guitar through spectrum-enriching effect devices such as boosters, fuzz boxes, doublers, etc. The same applies to other instruments generating more-or-less pure sounds. Once the carrier is rich in harmonics, it is possible to mold it into the timbral shape of vocalized sounds, sung or spoken into a microphone connected to the analyzer input. Now, dynamics and color are under complete vocal control and can be modified instantaneously due to the real-time nature of the vocoder.

On-stage use will require certain precautions, depending on acoustic conditions. Due to the phase shift introduced by the sharp filters, the vocoder can be very sensitive to acoustic feedback. Also, background noise can trigger horrifying effects. The classic trick with two reverse-phased microphones can be helpful, as well as directional, close miking. When close miking, attention should be paid with respect to inherent low-frequency boost, and when the microphone does not have an internal roll-off filter, conditioning with an external equalizer or by means of the attenuators at the synthesizer control inputs will be necessary.

Applications in the electronic music studio are almost unlimited, as long as the vocoder is an "open" system, meaning that the user will have access to everything controllable inside the unit. With the help of the analyzer section, frequency-dependent triggering or modulating effects can be obtained. A frequency-controlled band-pass filter is easily realized by sweeping a sine wave through the analyzer band. Interesting percussion effects can be generated by applying random sequences of sine bursts to the analyzer and by feeding pink noise into the synthesizer filter section. Depending on the positions of the patch pins in the matrix, intricate rhythmic structures by a multitude of percussive instruments can be obtained. In all, the vocoder can be used for a long list of effects which do not at all resemble the classic vocoder effect.

At least for the vocoder, the expression, "Your imagination is the limit," is not hype. ■

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MICHAEL RETTINGER

Reproducing Electronic Music in the Control Room

Real advancement in the control room means increased low-frequency recording capability—not increased gadgetry.

THE SOUND RECORDING business is a highly-competitive one. When a studio can offer blue-sky novelties like Plasmatronics loudspeakers ($6500 a pair), bass traps ($2000 each, if space is available in the studio), helium-filled balloons for greater sound dispersion along the ceiling (only $500 per balloon), the more-fickle clientele may become attracted to that recording facility.

But such devices are not really effective means of innovation or improvement; they are more in the nature of gadgetry and interesting exotica.

This writer believes that a *real* advance in sound recording and reproduction can be made by extending the reproducible low-frequency range of the program. There are at least two arguments in favor of this type of advancement: synthesizers can now generate bass notes lower than the more-traditional musical instruments, and, digital recorders can record and reproduce such notes with a much-greater signal-to-noise ratio than the usual analog recorder.

For good reproduction of low bass notes, the speaker's cone suspension must have a low resonance frequency. In addition, for large air-volume displacements of the loudspeaker diaphragm, the cone travel from its mid- or rest-position may often be more than 0.5 inches (1 inch total excursion). This tends to result in distortion which can readily be heard, because the ear's sensitivity increases with frequency in this range of the audio spectrum.

FIGURE 1 shows the loudness level contours below 100 Hz in a binaural free-field environment. For a loudness level of N phons, the sound pressure level (SPL) may be found by the formula

$$SPL = 160 - (67.5 - 0.4N)\log f.$$

As an example, at 25 Hz, when the loudness level is to be 100 phons, the sound pressure level is 121.5 dB, relative to 0.0002 microbars.

The cyclic air volume displacement of a circular piston in an infinate baffle may be calculated by the formula

$$V = \frac{0.0004(R)10^{SPL/20}}{f^2}$$

where

V = air volume displacement, in cubic feet-per-cycle,
R = listening distance, in feet,
SPL = sound pressure level, in dB, and
f = frequency, in Hertz (cycles-per-second).

With the frequency and sound pressure level given above, and a listening distance of 13 feet, the formula gives us a displacement of 1 cubic foot (1728 cubic inches)-per-cycle. At a frequency of 25 Hz, this means an air-volume displacement of 25 cubic feet-per-second.

Mr. Rettinger is an acoustics consultant and author of "Acoustic Design and Noise Control."

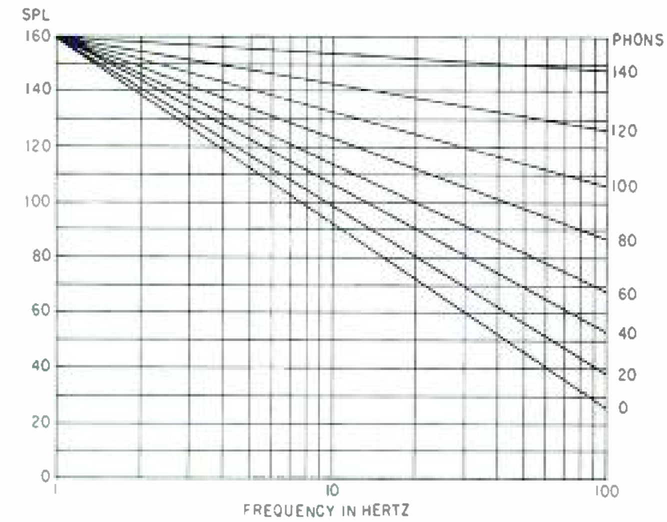


Figure 1. Idealized contours of equal loudness below 100 hertz.

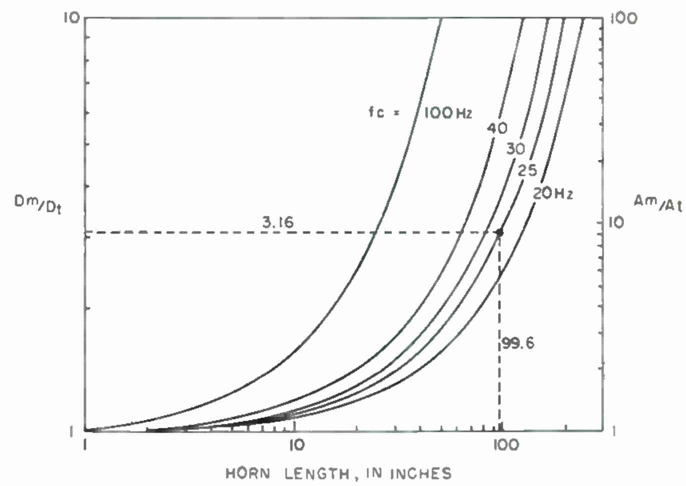


Figure 2. Horn length, in inches.

A loudspeaker with an 18-inch diameter cone has an active diameter of 16 inches, and therefore an active circular area of 201 square inches. Given a cone travel of 1 inch, we will require 8.6 speakers (1728/201) to produce the required displacement.

To avoid distortion in the reproduction of such low notes when the baffled diaphragm is driven to full excursion, an exponential horn should be used. Such an acoustic transformer can increase the efficiency of the reproduction system several times, thus requiring much smaller cone displacements (that is, fewer speakers) at full output. There is indeed a close relationship between speaker efficiency and distortion, particularly when impedance matches are involved.

An exponential horn acts as an impedance-matching device, acoustically coupling a heavy cone to the relatively-light air medium. It is not unlike a step-down transformer, coupling a high-impedance source to a low-impedance load. When the ratio of the horn's mouth-to-throat diameters is 3.16:1, the sound pressure at the mouth—like the high voltage at the amplifier output—is decreased by 3.16. However, the volume of air flow at the mouth—like the current through the loudspeaker coil—is *increased* by 3.16. When the horn is well made, there is no power loss along its length by acoustic energy absorption in the horn walls. The acoustic power along the horn length remains constant, because it is given by the product of sound pressure and air particle velocity, which remains constant. Note that a ratio of 3.16 for horn mouth-to-horn throat diameter corresponds to a ratio of 10 for the relative areas of mouth and throat, as long as the cross-section of the horn is either round or square.

How long must an exponential horn be, which is to have a cut-off frequency of 25 Hz and a diameter ratio of 3.16? The equation is

$$L = \frac{\log D_m - \log D_t}{0.217m}$$

$$= \frac{414.9 \log(D_m - D_t)}{f_c}$$

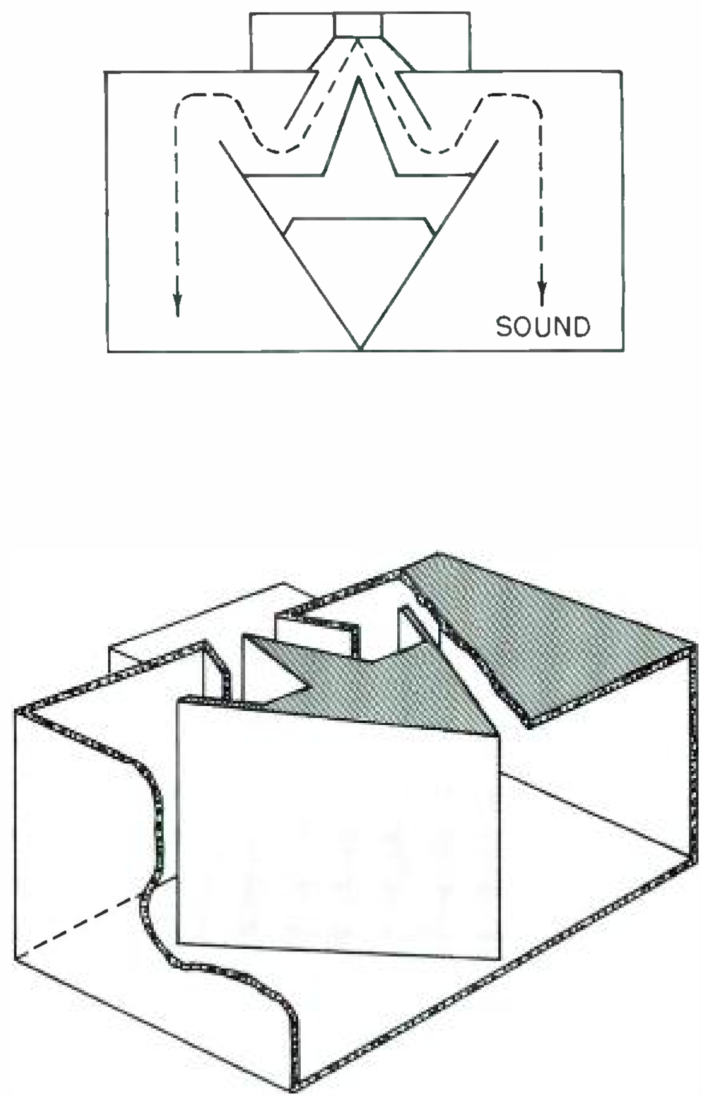
$$= \frac{414.9 \log(3.16)}{25}$$

$$= 8.29 \text{ feet}$$

(This formula is explained in detail in the author's "Restricting Sound Radiation" in the October, 1980 issue of db—Ed.)

FIGURE 2 is a graph of horn length (L)-versus-diameter ratio (D_m/D_t), for various cut-off frequencies. The right-hand vertical axis also gives the mouth-to-throat area ratio A_m/A_t).

Figure 3. A folded low-frequency horn.



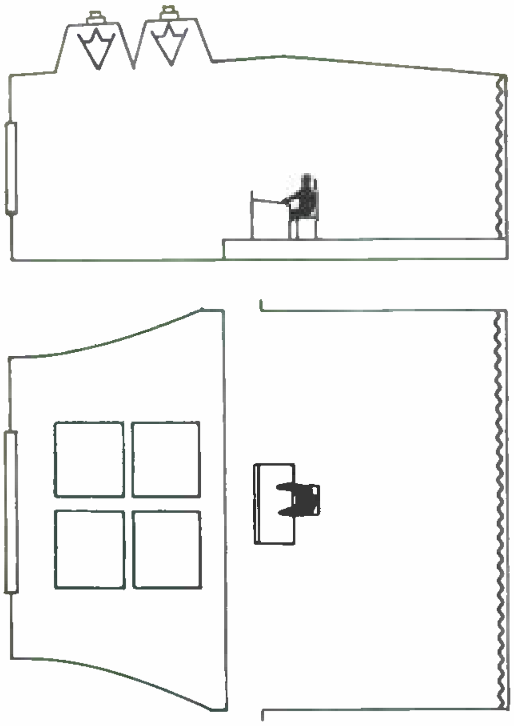


Figure 4. Low-frequency horns installed in Control Room ceiling.

A horn with a length of about 8.3 feet (99.6 inches) would probably have a total length of 10 feet, once the woofer is installed at the throat. To shorten this dimension, a folded horn may be used. FIGURE 3 shows such a unit, as was employed for the reproduction of those very low notes in Universal Studios' "Sensurround" system. The system is intended to reproduce not only these low-frequency sound waves, but also wind rumbles and other below-25 Hz effects.

FIGURE 4 shows the floor plan and elevation of a recording studio control room designed to accomodate the horn system just described. To facilitate construction, the horn mouths may be either square or rectangular. Because of the frequencies that are reproduced, these units are practically omni-directional, and need not be installed in the front wall of the room.

If digital recording technology eventually allows us to routinely record very-low frequencies, perhaps the audiophile listening room of the future will also see the installation of such horn systems, when (and if) these sounds make their way to the lp record. ■

References

The best sources of information about loudspeakers and sound reproduction are the books by H. F. Olson. His first book "Applied Acoustics" has become a rarity because it is out of print. It was originally published by P. Blakiston's Son & Co. Inc., Philadelphia. It was written in collaboration with Frank Massa, a Swope Fellow at MIT. Two other books by H. F. Olson are "Acoustical Engineering" and "Modern Sound Reproduction," both published by Van Nostrand Co. Inc. in Princeton, New Jersey.

HAL LION and JIM FOX

Metric Music

In a series of radio news features introducing America to the metric system, what could be more appropriate than having "metric music" playing in the background?

THE U.S. METRIC BOARD has the unenviable task of introducing the general public to the "delights" of the metric system. So, when the agency told us they wanted background music for a series of radio news features, we asked ourselves, "What does a ten-tone musical scale sound like?" We decided to find out.

Since the metric system divides everything into tens, we reasoned, why not go all the way with this thing and make the music metric too? After all, electronic technology being what it is, there's no reason to stay tied to the traditional twelve-tone scale; the question was, where to get some guidance for this excursion into the unknown? We found the frequencies of the conventional scale in The Audio Cyclopedia, but did not see any explanation as to where these numbers came from. So, we got out the log tables and the calculators, and with the help of a little algebra devised our own formula. It worked very well, and we later found it verified in a textbook chapter on musical scales.

If you've forgotten—or never learned—the mysteries of musical scale construction, here's how it's done.

The musical scale is a geometric progression. For example, the lowest A on the piano is 27.5 Hz, and the A above it is 55 Hz. The one above that is 110 Hz and so on through 220, 440, 880, 1760, and 3520 Hz. In other words, to find the frequencies of any series of octaves, just keep multiplying by 2. It follows that to find the frequencies of the notes on the scale within the octaves, you need to multiply by a figure that will, for each frequency in the scale, give the frequency of the next succeeding note. For the conventional 12-tone, equal-tempered scale, that figure is $\sqrt[12]{2} = 10^{\log 2/12} = 1.0594631$. When any number, f, is multiplied twelve times by 1.0594631, the final product will be 2f.

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To illustrate, using the octave between A-440 and A-880, begin by multiplying 440 by 1.0594631. This will give you the frequency of A#. Then multiply A# by the same number to get the frequency for B, and so on. The result is as follows:

A	440.000	E	659.255
A#	466.164	F	698.456
B	493.884	F#	739.989
C	523.251	G	783.991
C#	554.365	G#	830.609
D	587.330	A	880.000
D#	622.254		

Using the same principle for our ten-tone scale, we multiplied each frequency by $\sqrt[10]{2} = 10^{\log 2/10} = 1.0717735$. Starting with 440 Hz, this metric musical scale would be:

0	440.000	6	666.915
1	471.580	7	714.782
2	505.427	8	766.084
3	541.704	9	821.069
4	580.583	0	880.000
5	622.254		

Notice that Note 5, at the mid-point of the metric scale, has the same frequency as D#, at the mid-point of the conventional scale.

But rather than start with 440 Hz, we wanted an authentic, metric basis for our scale. We assumed a day divided into 100,000 "metric seconds" instead of the 86,400 conventional seconds and established a base frequency at 10 cycles per "metric second" (cpms). Then, of course, we had to make a concession to the fact that all of our equipment is calibrated in standard seconds. A quick calculation will show that 1 second = 1.1574074 "metric seconds," so that 10 cpms = 11.574074 Hz.

Beginning with this sub-sonic frequency, we kept multiplying by 2, until we reached the two octaves between 185.185 Hz, 370.370 Hz and 740.740 Hz (corresponding to 160, 320, and 640 cpms) that were within a tonal range suitable for our purpose. The metric octaves ("dectaves"?—Ed.) up through the range we used are as follows:

cpms	Hz
10	11.574074
20	23.148148
40	46.296296
80	92.592592
160	185.185185
320	370.370370
640	740.740740

These tones, incidentally, correspond almost exactly to F# on the conventional scale. The middle tone in the octaves we chose to work with (370.370 Hz) corresponds to the F# above middle-C (369.994 Hz). Having identified the octaves, we then used the procedure outlined above to make the 10-tone scale.

The next step was to find a way of producing these sound intervals on available equipment. For this task, we turned to our resourceful synthesizer player, John Abernathy. He set the VCF (voltage-controlled filter) of his Arp Odyssey synthesizer to self-oscillate. He then patched the keyboard into the VCF and, taking a range of twenty notes, adjusted the VCF attenuator to "squeeze" them, so to speak, into an octave. When this was done, middle-C and the G# above C-above-middle-C sounded just one octave apart. This adjustment actually produces a 20-tone scale, from which the 10-tone scale can be realized by playing every other key: C, D, E, F#, G#, A#, C, D, E, F#, G#. If metric music ever catches on, one can of course devise an appropriate keyboard, perhaps with some color coding to distinguish octaves and half-octaves (Notes 0, 5, 0, 5, etc.).

When the scale had been established by adjusting the attenuator, the VCF cut-off was adjusted to bring it into line with the desired base frequency. A sine-wave audio generator was used, first to establish the base frequency, and then to check the other tones in the scale, verifying the intervals as calculated, and *voilà*, "metric music"!

Now comes the hard part—composing an acceptable melody on this admittedly weird scale, which of course has no harmonics to speak of and needs to be heard in a sequence of single tones. John Abernathy tried a number of sequences. For the most part, attempts to adapt familiar melodies from the conventional scale did not yield very interesting results, but he composed some melodies that used the strange intervals to advantage. We finally settled on four melodic segments of five notes each—since the number 5 seemed appropriately metric. These segments are noted below on the special ten-tone staff we have designed.

We put these segments together in various combinations, producing a melody that was then surrounded by some other interesting musical effects. The Metric Board accepted the final result and uses it in the background of "Metric Magazine," a series of radio news features that has been distributed to stations throughout the country. ■

RALPH HODGES

Zoop, Beep, Brap, Broop

Welcome to the brave new world of digital synthesizers.

IT MAY SEEM like yesterday when *Switched-On Bach* was climbing the charts, but it's actually ancient history. Bob Moog is reported to be breeding horses in New England, Tomita is presumably dancing with snowflakes somewhere in Hokkaido, and the days in which you punched up a sawtooth wave from a glorified audio generator, added a little square for seasoning, and then ran the whole thing through a ring modulator are dead and gone. The generation of electronic music is overwhelmingly digital by this time, and even if you believe that this development is hazardous to your health and vitality, you cannot deny the logic and potency of the approach.

Actually, there are probably not too many things worth doing that an analog synthesizer could not somehow be designed to accomplish; and with luck and care, it might even be possible to keep noise build-up from swamping the composite. But the fact is that things are easier with digital in so many ways. There is hardware flexibility; a typical analog circuit may be limited to one function, whereas a digital circuit can often be persuaded to perform several. There is control interface; the sound-generating portion of a digital synthesizer tends to be computer-like, just as the control section is computer-like. They live in the same sort of time frame, have similar signal-handling characteristics and drive requirements, and they present the overall designer with only one discipline to master fully, instead of two. And of course there are the matters of cost, complexity, compactness, and portability, which only increase the probability that anyone involved in audio work is going to have a run-in with one of these contraptions sooner or later. What is this brave new world going to be like?

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THE NATURE OF THE BEAST

First of all, what *is* an electronic-music synthesizer, really? (For this discussion, we'll restrict the category to *digital* synthesizers, they being in the majority and very much on the rise.) Is it a musical instrument, a signal processor, a sound-effects library, an automated music-composition machine, a recording device, a multi-media effects coordinator, a glorified mix-down engineer and editor, or a mindless brat of an electronic nuisance? Today, it's likely to be at least a little bit of all these. Tomorrow, it will probably take on the scheduling and billing for sessions and arrange for the contractor to come in and clean out the fish tank. The device need have no real identity whatsoever. The specific functions it can routinely perform have, as a rule, been deliberately circumscribed by the designer, who is coming to realize that normal people have to have some limitations to fetch up against, lest they get lost in potentialities and intricacies.

As a beginning, let's look at a comparatively simple but not especially typical device: the LM-1, offered by Linn Electronics for a low-key (in this league) $5,000. The LM-1 specializes in something that may come as a bit of a surprise: real music, and specifically real drum and percussion music. Virtually every sound it is capable of making originated with a real instrument and a microphone, the pick-up of which was digitized and stored in hard read-only memory as single beats of a kick, a snare, two toms, two congas, tambourine, cabassa, high-hat, cowbell, and a little bit of *et cetera.*

The machine puts these single recorded beats together in the repetitive pattern of your choice, working basically in—but not limited to—two-measure chunks of 4/4 time. Select any one of a hundred random-access memories on the numeric keyboard, hit record and play, then whack out whatever you please on the pushbuttons assigned to the individual instruments, in time to the metronome beat the device will give you (at whatever speed you want, of course). Want more instruments than your two fingers can play at once? Store what you've got, repeat the original recording sequence, and, in effect, overdub. You need another time signature? Count it out as if you were playing the actual instrument, and the memory will give it back to you on demand. Your problem is that you can't play at all? Well, if you can manage even a crude approximation of what you're after, the LM-1's time base will correct your timing errors, to the nearest 1/16th note as a matter of course, but to 1/32nd-note triplets if you elect.

After you've finished loading up two measures with as much racket as you can tolerate, what then? A chain function enables you to link up memories for a continuous rendering of perhaps ten minutes of fairly complex, ever-changing rhythm patterns before storage capacity is exhausted. You just keep punching the new patterns in, or drawing on those that have already been stored. Should you need still more storage, the LM-1 is based upon control signals that even a lowly cassette deck can record and play, so these can go right out a jack on the rear panel and get stored on tape.

Needless to say, the LM-1 will give you back anything you've stored on demand, at the press of a few buttons. It also lets you hop quickly around from memory to memory, changing or erasing anything you wish. But in common with the more complex devices we're getting to, it is not intended to duplicate the functions of a multi-track console. If you require some reverb, you must get the signal out and into an echo send and return. There are separate outputs for all the instrumental sounds provided, but they are intended to be panned, balanced, and equalized on the board (which incidentally might take up more inputs than your usual scheme of miking a real drum set). The outputs, like the signals themselves, are mono, if that makes a difference to you. On the other hand, the various percussion sounds can be continuously varied in timing and pitch independently—something that only technology like this can manage.

SCIENCE-FICTION LAND

Useful as the Linn device can be, it does not really qualify as a synthesizer, being limited and ultimately inflexible in the variety of burbles and bleats it can emit. The big jobs, coming in at approximately $14,000 and going up rapidly from there (especially with options), can generate any sound you've ever heard or can imagine, render it up as rock, boogie woogie, or a Viennese waltz, play it forwards or backwards, alter it even further beyond comprehension as they do so, and even print it out on a sheet of music paper. They come with organ-type keyboards, standard or optional, and therefore are conveniently playable in real time. What can be achieved at the keyboard will give some idea of the flexibility available. Key velocity—the speed or force with which the key is struck—can be programmed to regulate parameters that include dynamics, attack, vibrato, glissando, portamento, sustain, damping, or a combination of these. The mere act of pressing a single key can cause a note to rise slowly, blossoming with a peacock's tail of unexpected overtones as it comes, and then decline abruptly to sustain with a more conventional harmonic structure but a sudden addition of vibrato, and finally fade away into an eerie-seeming-silence. At a demonstration of the Con Brios ADS 200, a quick tap of a key brought forth a clear, mellifluous bell tone that decayed naturally; sustained pressure on the same key made it die abruptly and dissonantly, as if the clapper had remained pressed against the bell. With a new program punched in, the same key provided only the raspy squeak of a saxophone-reed attack with a quick tap, but the full bloom of the instrument when the key remained depressed. In other words, totally opposite decay modes for the same playing action. But I suppose it makes a certain amount of sense, once you get to thinking the way synthesizer people think.

The Con Brio synthesizer really does work hard at relating to the uninitiated player programmer. It is almost completely integrated into a single chassis, with a two-rank organ keyboard below, a control panel to the upper right, and a CRT and floppy-disc drive on the upper left. (Additions, many of them optional, include foot switches and pedals, an alphanumeric keyboard terminal, a printer that raps out a score of what you've done in conventional notation, a sub-chassis for additional hardware, and continually floppy-disc software.) The controls are intentionally labeled for comprehension by an analog person, and many of them are continuously variable knobs. Pushbuttons are illuminated only when the time has come to use them, and the CRT display makes an admirable attempt to keep you aware of what has happened through each step of the programming.

The Con Brio 200 is based on digital oscillators—64 of them (expandable to 256), which is said to be the current record number. Digital oscillators put out discontinuous pulses that are ultimately smoothed into a continuous waveform by a low-pass filter, but otherwise they present no obvious difficulties *vis-a-vis* analog oscillators, and they confer a lot of nice advantages. But how do you go about manipulating them?

Many will select a combination of the 160 preprogrammed wave patterns from the two floppy discs supplied by the manufacturer, and just begin making music. But the courageously adventurous user will attack an earlier point of the input. Basic tones can be built up from sine, square, triangle, sawtooth, pulse, and noise functions of the oscillators, singly or in mixtures. These are pushbutton-selected. Then additional oscillators are brought in for control signals to modulate amplitude, relative phase, frequency, and to provide additive spectral touches, with the CRT display acting as an envelope guide along the way. The end result is called a "voice." It's a hypothetical musical instrument, with its characteristics stored on floppy disc. Like the preprogrammed voices, it is indexed by number and can be called up for playing on the keyboard at the touch of a few numeric buttons; and it will never go away unless you choose to erase it.

But now it is necessary to play the instrument, and for that purpose a properly instructed keyboard musician is certainly

Figure 1. The Con Brios ADS-2000 Digital Music Synthesizer.

the best alternative. Besides the voice constructed above, as many as three others can be accommodated on the two-rank keyboard at any one time. And the keyboard can be assigned, note by note, to any one voice, so if you seek the ultimate, you divide it electronically into four sections, apportioning the necessary number of keys to each. Then you figure out what effects (as described above) you want to be handled by such factors as key velocity, and what are better controlled by foot switches and pedals. The basic ADS 200 has two outputs, so the performance can be panned as it plays. Perhaps more importantly, the action of the foot switches can be programmed to sequence in any of the other voices stored in disc memory. They pop up on the keyboard with a speed that is one of the most attractive features (and a trade secret) of this manufacturer. This capability is labeled "Ensemble" on the synthesizer's control panel, and eight buttons do it all, plus a good deal more that there's no space to describe.

Once the performance has taken place (and of course you don't need Horowitz to perform it; you can go back, patch up, alter, make insertions, and indulge in other almost obscene liberties), it is directly retrievable from the floppy disc that received it, whether it was simultaneously sent out to a studio recorder or a live audience or not. You can get it printed out as parts on charts, and you can transfer any part to another voice, to see if it sounds better. You can alter it in pitch and overall tuning to agree with other instruments that might be mixed with it. The same goes for tempo.

THE VALLEY OF THE DOGS

The Fairlight CMI is not based on digital oscillators, but employs a free-floating record/reproduce system that is coordinated by a variable sampling rate. Innate storage capacity, in seconds, depends crucially on the complexity of the input, but most of the storage takes place on floppy disc anyway, so there is no cause for concern.

Two disc drives are incorporated in the main CMI unit. One holds the disc that regulates the control functions of the system, and is a permanent fixture (ROM, in other words). The other is the field on which you play your games.

One of the more impressive games the CMI can play is the recording of a few seconds of any sound or sound-like electronic signal, which can then be altered as desired and thrown onto a keyboard to become music. When I visited the Santa Monica distributor for this Australian-made device, there proved to be some dog barks stored in memory for experimentation. And since Santa Monica is not too many feet above sea level, and many surrounding parts of Los Angeles are . . . well, valley of the dogs.

This is not at all meant as a denigration. The Fairlight CMI is a fascinating creation. But like the Linn LM-1, it probably doesn't qualify as a true synthesizer, because it is silent unless it has received an input at one time or another. Once it has the input, however, the things it can accomplish are truly arresting. A waveform appears on a CRT screen (625-line base) and you put a light pen to it to perform virtually any alteration you like. This is just the beginning. As you begin to shape the dynamic and harmonic evolution of that waveform over time, the CRT display assists with readouts of up to eight harmonics graphed as amplitude-against-time, 3-D representations of spectral alterations with decay, bar graphs (alterable also by light pen) of relative overtone strengths, and a wealth of codes, instructions, and sequencing signals. You can literally make the smashing of a beer bottle sound like an operatic soprano, if you're willing to work at it and to develop the sense of eye-ear coordination that seems to come from experience with this technology. Furthermore, the CMI would seem to offer the chance of replicating, convincingly, any sound you could come across. It does not just switch and stack oscillators, but instead

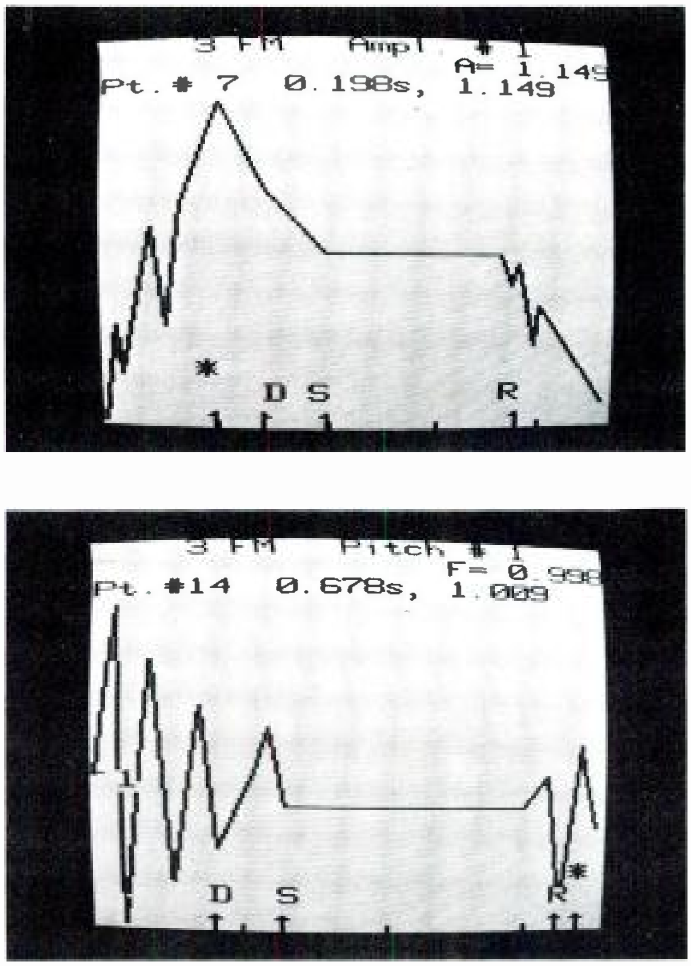
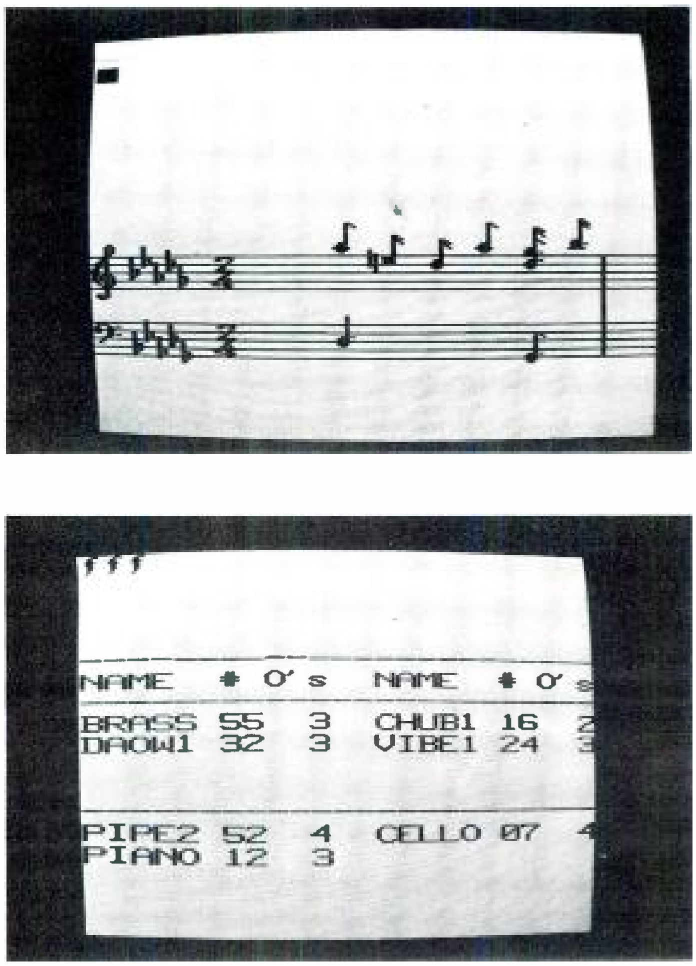


Figure 3. CRT displays of amplitude (A) and frequency (B) envelopes.

Figure 2. The Con Brio's CRT performs a variety of functions, including (A) the display of real-time notation for keyboard performance, and (B) a read-out of digital files available to the user.



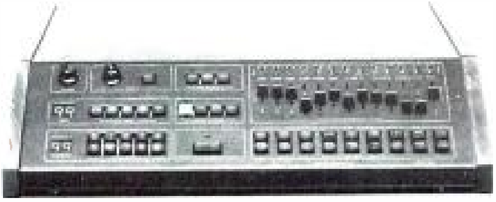


Figure 4. The LM-1 Drum Computer

gives you intimate control over the actual waveform, which means that you can adjust for the vagaries of relative phase, studio/room acoustics, unknowns in the aborning science of psychoacoustics, and all the rest. Approaching the project as a programmer, it might take you a year to work your way through even one movement of the Bartok Fourth String Quartet as performed by the ideal ensemble, but in theory you could do it.

BACK TO THE RANCH

There's no need to point out that the Fairlight CMI potentially provides all the storage, retrieval, and performance capabilities expected of a computer-based synthesizer, and as handily as most of the rest. But larger questions loom: What if one of these things actually turns up at your studio door? What if the musician/engineer/programmer drops dead in the middle of a session? What if he thought he knew how to retrieve all inputs but has suddenly discovered that he doesn't?

There really is no question about what to do. Call a person as close to the manufacturing source as possible. Waste no time with local technicians, who will have to spend hours, or even days, puzzling out the control logic before they can begin to do useful work. These things are new and unknown in the field. If minutes count, allow 30 of them for experimentation and then get the advice of someone who really knows. The practical synthesizers have safeguards against the loss of vital information, but those very safeguards will block you from haphazard access. And because a modern synthesizer may very well arrive with the best part of a rhythm track, if not more, already loaded up in memory, extreme care is the best policy.

Should you otherwise be afraid of synthesizers? The best advice I've gotten indicates not. In the hands of an extremist, a synthesizer can zap you with a very heavy level of near-ultrasonics before you can catch your breath. But the digital synthesizers could not go truly ultrasonic even if they wanted to, because of low-pass filters at their outputs that limit them to 16 kHz or less. And no one is much interested in sound compositions that rise that close to extreme levels at inaudibility anyway. Modern synthesizers have too much to offer in the middle of the audio band, so there's no reason to press beyond it.

Digital synthesizers are usually set up so that they can lock onto an externally-generated time code, or impose such a time code on external equipment. They are quite flexible, and generally work well with such a universal standard as SMPTE time code. If you have any doubts about a particular interface, a few quick questions to an authoritative source should clear them up right away. Beyond that, I have no more advice about synthesizers except for a statement by Roger Linn: "Have fun with them. They're made to make people happy." ■

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The 67th AES Convention

MORE THAN 6000 visitors jammed New York's Waldorf-Astoria Hotel last fall for the Audio Engineering Society's 67th convention (31 October-3 November, 1980). In addition to the traditional fare of technical papers and equipment exhibits, the 67th featured a nine-session workshop series which included digital editing, video for audio, and microphone techniques. And, to help answer the recurring question, "How do I get started in professional audio?", one of the workshops was an educational fair, at which "A gathering of representatives of universities and educational institutions offering courses in audio (were) on hand for personal discussions with prospective students. Information (was) provided on entrance requirements and curriculum, with details of special courses available of interest to those entering or involved in the audio field." (From the AES convention program. For a copy of the AES's Directory of Educational Institutions, write to the Audio Engineering Society, 60 East 42nd Street, New York, NY 10017.)

The convention saw the introduction of another volume in the AES's anthology series. Edited by Gotham Audio president Stephen Temmer, the 520-page volume is entitled "Disk Recording—Volume 1: Groove Geometry and the Recording Process." This brings the AES anthology series up to five volumes. The other four are: "Loudspeakers," "Microphones," "Quadraphony," and "Sound Reinforcement." The anthologies are available from the AES at $22 a volume ($19 for AES members).

THE EXHIBITS

179 companies jammed the hotel's grand ballroom and its balconies, several large meeting rooms, and two floors-worth of guest rooms. It has become somewhat of a tradition for at least one company to pull off a little surprise by unveiling a totally-new product at the show. This time around, it was MCI's turn—almost. Almost, because the Fort Lauderdale-based company brought along its first PCM tape recorder, but kept it in a secluded location where it was shown by invitation-only. Needless to say, we weren't invited, so all we can report is that a small quanity of machines have already been shipped to a handful of clients here and abroad. For more details, contact MCI directly.

Otari, not quite as shy, showed off its new pro' MTR-10 series of two- and four-channel recorders on the main ballroom floor. Replaceable head stacks allow the four-channel machine to be converted to two-channel operation on either quarter-or half-inch tape. The "bells-and-whistles" include a dual real-time counter, internal multi-frequency sine/square wave generator, optional tape locator with ten-postition memory, and, of course, a micro-processor based servo transport system. Projected deliveries should begin at about the time you read this convention report.

At Panasonic, projected deliveries of the company's latest black box are still off in the future somewhere. Ambiguously dubbed a "Localization Processor," this second-generation prototype jolted at least a few of the golden-ears crowd, as they casually manipulated the quad pan-pots on the processor's front panel. What's so exciting about a quad pan-pot? In four-channel sound, its certainly no big deal, but, in *two*-channel systems...? It turns out that with only two output channels, the Localization Processor creates clear images in the entire frontal half-plane. Having followed the R&D work on the project for several years, this reporter found it great fun to watch people search for those side-placed phantom speakers, which of course didn't exist! (For an early report on this project, see "The Shape of Things to Come" in the June, 1979 **db**—Ed.)

OTHER SIGNAL PROCESSORS

At Gotham Audio, a second-generation digital reverberation system—the EMT 251—was demonstrated. If you liked R2D2, you'll love the EMT 251, which—as Gotham points out—retains the "space machine" look of the earlier model 250. But beauty (?) is only skin deep (better make that "front-panel deep"). Underneath, it's all business, with twice the memory capacity of the 250, plus self-diagnostic and internal error correction systems. The 251 provides three delay lines, a reverb cluster just prior to the onset of reverberation, and the reverberation program itself. All this is assignable across the stereo output.

The 251 is compatible with console automation systems such as the Harrison Auto-Set, and others. In other words, control-setting variations may be recorded on a data track or floppy disk, along with the usual console fader information. Hollywood's Sunset Sound Studios has taken delivery of the first 251, but you can have your own for just $20,000.

The dbx 900 series is a modular signal processing system which now comprises five plug-in components. These are: a de-esser, a compressor, a noise gate, a parametric equalizer, and a flanger. Additional modules are on the drawing board, and up to eight will fit in a 5½-inch rack mount mainframe.

Briefly summarizing the 900 series features: the 902 de-esser can be used either as a conventional broad-band processor, or as a high-frequency attenuator; the 903 compressor features the dbx "over easy" system, and provides compression ratios from 1.1:1-to-∞:1; the threshold range of the 904 noise gate is from -40 dB to +10 dB, with attenuation continuously variable over a 60 dB range; the 905 flanger provides delays up to 40 ms, for flanging and doubling effects. In addition, there are front-panel controls for feedback, dry/delay mix, depth, sweep speed, and phase reversal.

In 1969, Allison Research introduced the original KEPEX (KEyable Program EXpander). The KEPEX II is now manufactured by Valley People, Inc.—the recently-formed company that merged Allison with Valley Audio. K-II's expansion rate is variable from 1.1:1 to 100:1, with an attenuation range of up to 100 dB, and a threshold that is variable between -40 dBv and +20 dBv.

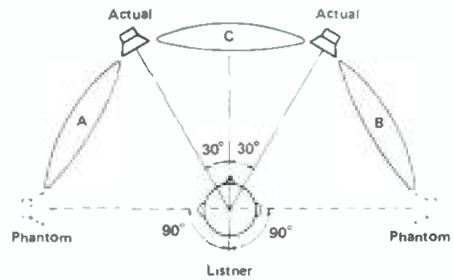
Gain Brain II is an update of the earlier Allison model 700 Gain Brain. In addition to the customary limiter/compressor functions, GB-II features a switchable linear/logarithmic release time, and a "ducker" function. This allows up to 48 dB of attenuation in the normal signal path. The attenuation is regulated by the presence of a signal in a side chain. Thus, a narrator's voice can be heard over a music program, or an interactive vocal-instrument balance can be set up.

Valley People also offers consultation services, including LEDE acoustical design and TEF measurements. (LEDE and

Otari's MTR-10 series of professional tape recorders is available in two- and four-channel formats. The MTR-10-4 is seen here.

Panasonic's second-generation prototype Localization Processor. Once in production, the control section will be a considerably smaller package, with most of the electronics located in a remote rack mount location.

The Localization Processor gives the operator control over panning assignments within the entire frontal half plane. This includes depth, as well as angular positioning.



TEF are registered trademarks of Synergetic Audio Concepts.) In addition, the company provides systems engineering, installation and construction services.

AT THE MIKE SHOP

"The Great British Spring" has nothing to do with the weather: it's a six-spring reverberation system available at the Mike Shop with ($624) or without ($499) a two-band parametric equalizer. The Mike Shop also markets the BEL BC3 Noise Reduction System. According to the spec sheet, the BC3 is dbx-compatible, with separate encode and decode circuits, thereby eliminating the need for record/playback switching. An eight-channel system costs $1,575.

The Mic Mix Company's new XL-500 Master Room is an elegant three-piece reverberation system. The main control system is a 5½-inch rack mount unit, containing left- and right-channel equalization, chamber mode selector (plate, room, hall) and decay and mix controls. Chamber mode and decay time, as well as equalization bypass, may also be adjusted from a small remote control box located near the operator. The remote control is activated by depressing any of its mode switches. Decay time and chamber drive readouts are located on both the main control unit and the remote controller.

MICROPHONES AND ACCESSORIES

AKG's new C-414EB/P48 is being publicized as a "studio microphone for digital recording." Although its output is still analog, AKG states the 414EB is equal to the "pristine transmission qualities of digital technology." This means an equivalent noise level of 18 dB SPL (DIN 45 405), and a dynamic range of 124 dB. At 142 dB SPL, the THD is 0.5 percent. A four-position switch selects omni-directional, cardioid, hyper-cardioid, or figure-8 patterns.

Audio-Technica has introduced six new electret microphones and a battery power supply. Four of the mics are in A-T's "Artist Series": the ATM 10R (omni) and the ATM 11R, ATM 31R and ATM 91R (uni). The spec sheets identify the 10R and 11R as instrument mics, and the 31R and 91R as vocal mics. In Audio-Technica's regular series, the AT 803R is a sub-miniature omni-directional design, and the AT 813R is a standard-sized uni-directional microphone.

Panasonic's expanding RAMSA series now includes the WM-8100, a uni-directional back-electret condenser microphone, with low-cut, high-cut and sub-sonic filter switches, and a floating suspension system. A built-in LED indicates battery voltage condition. When an external power supply is connected, it overrides the internal battery.

Audio Engineering Associates' MS-38 Matrix Decoder is used for recovering left- and right-channel information from M-S (Mid-Side) stereo signals. The decoder accepts line-level M-S inputs, and is designed for insertion in the signal path immediately after the microphone preamplifiers. Alternatively, it can be used for decoding two-channel M-S tapes.

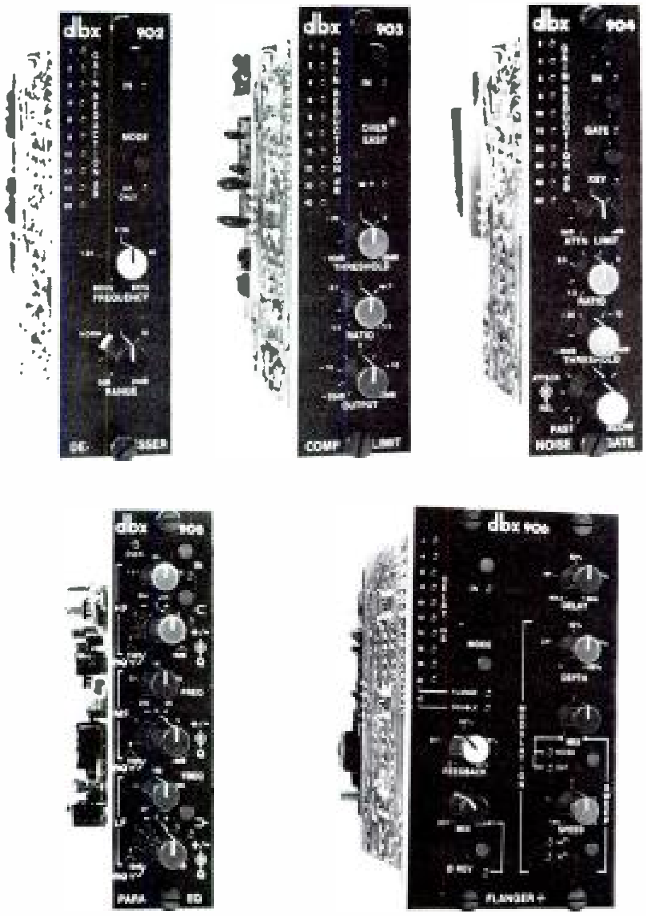
And back to the Mike Shop again. In addition to selling Britain's Calrec microphones, the company imports Musiflex mic cable. The cable is sold in 100-foot lengths and is available in eleven colors. A conductive thermoplastic shield takes the place of the customary braided wire or metal foil, which greatly simplifies the task of preparing the cable for plugs.

ODDS-N-ENDS

FAX Audio's Series 1 programmable fader combines a series of eight recessed membrane switches, with a Penny and Giles conductive-plastic linear-motion fader. A vertical column of 30 LEDs simultaneously indicates the levels of both read (dim LED) and write (bright LED) functions. The membrane switches control the read, write and update modes, as well as trim, mute and solo functions. A seven-segment LED displays group assignment, which is also controlled by a membrane switch.

Eventide Clockworks real-time spectrum analyzer is a plug-in card system for use with a Commodore PET, Radio Shack TRS-80 or Apple computer. For under $600, it turns your personal computer into a spectrum analyzer, with 31 one-third octave bands. The display can be either linear or logarithmic, and decay and gain are varied by keyboard commands. ■

The EMT 251 second-generation Digital Reverberation Unit.



The dbx 900 series of signal processors.

Audio-Technica's sub-miniature electret condenser microphone.

The WM-8100 is the latest addition to Panasonic's RAMSA line of pro' audio products.

This Audio Engineering Associates photo shows the MS-38 Active M-S Matrix System. Also seen is AKG's C34 stereo microphone.

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db People/Places/Happenings

• More than 400 commercial radio stations and radio-related companies joined the **National Radio Broadcasters Association** during 1980, according to **Abe Voron,** NRBA executive vice president. This boosts total membership in the full service radio-only organization to more than 1500. The 412 new NRBA members include independent and group-owned stations from 47 states, the District of Columbia, Puerto Rico and Canada. Of the 342 new commercial stations, 149 are independent and 193 group-owned, while the 70 new associate members are radio-related manufacturers, distributors, suppliers, consultants and educational radio stations. States attracting 10 or more new members were: Alabama, California, Florida, Illinois, Indiana, Massachusetts, Michigan, Mississippi, New York, Ohio, Pennsylvania, Texas, Virginia, and Wisconsin.

• **John R. Saul,** President and founder of **MICMIX Audio Products, Inc.** died on January 2, 1981 at the age of 49. Mr. Saul was a member of the **Audio Engineering Society, National Association of Broadcasters, Society of Broadcast Engineers,** and the **Society of Motion Picture and Television Engineers.** Mr. Saul worked as a senior project engineer for **LTV Corporation** and resigned in 1972 after 20 years of service. In 1972, MICMIX Audio Products was incorporated, and Mr. Saul assumed the position of president.

• Late this past December, the Los Angeles **Audio Engineering Society** Chapter held an informal meeting for its members with **Mr. Bill Putnam.** Mr. Putnam, chairman of the board of **United Recording Corporation,** reflected on the early days of the record business. His talk included how he recorded some of the music giants of the past: **Ellington, Sinatra, Patty Page,** etc. Putnam fielded many questions from the audience and showed the first module from an early recording console he designed. Recognized as one of the recording industry's founding fathers, Bill Putnam was responsible for the creation of United Recording, now **United/Western** located in Hollywood; the formation of **U.R.E.I.**—an industry manufacturing leader in studio equipment; **Coast Recorders** of San Francisco, and an information systems firm named **Teletronix.**

• **Empirical Audio** of Ossining, NY, is pleased to announce the sale and installation of a new **Trident** TSM Automated Recording Console to **MasterSound Productions,** of Franklin Square, NY. According to **Winn Schwartau,** president of Empirical Audio, MasterSound's Trident console is the largest board delivered to date. The custom TSM has 48 inputs, with 32 monitors, giving the board a full 80 channel remix capability. MasterSound is automating the console with a **Valley People** 65K programmer, and feeding the Trident console with **Ampex** MM1200 24 and 16 track tape machines.

• The **Recording Industry Mangement** program at **Middle Tennessee State University** has purchased a **Harrison Systems** 2824 series console. The console, along with an **Allison** 65K programmer, was installed in the RIM studio by **Studio Supply** of Nashville. Recording Industry Management is a four year degree program covering all aspects of the recording industry.

• **Community Light & Sound** has announced that **Chris Foreman** has joined them in the newly created position director of marketing. Mr. Foreman, the former director of applications engineering at **Altec-Lansing Corp.,** is a member of **AES, IEEE, SMPTE** and is a regular contributor to many audio magazines and trade journals.

• **Charles P. Repka** has joined **RCA** VideoDisc Operations Division in Indianapolis, Indiana as a member of the Systems Engineering group. His experience includes work as a Systems engineer on the **Sky Lab** program at the **Marshall Space Flight Center** and as a EW Systems engineer for **I.T.T. Avionics.** In recent years, Mr. Repka has been a recording engineer for **Vanguard Records,** disc mastering engineer for **Masterdisk,** freelance recording engineer and writer on audio topics for a variety of audio journals. Mr. Repka has been an active member of the **AES,** serving as committeeman, vice-chairman and chairman of the N.Y. Section. He has also chaired the Disc Technology session at the N.Y.C. AES convention.

Mr Repka's duties at RCA will include the development of audio system parameters for the current and future Videodisc Systems.

• **Empirical Audio** has completed negotiations with **Ran-Steele Audio** of New York City for the sale of 24 **Advanced Music Systems** Disc Mastering Digital Delay Lines. Ran-Steele Audio is a manufacturer and supplier of custom mastering facilities world-wide. According to **Tom Steele,** president of Ran-Steele, the use of Disc Mastering DDL's will soon replace the conventional cutting method of using advance heads to feed the cutter head preview signal information. The AMS DDL is a programmable stereo unit, with up to 1.3 seconds of delay, and a bandwidth of 27 kHz.

• **Compact Video,** Burbank, Calif.; **Rodel Audio,** Washington, D.C.; and **ITV Ltd.,** Edmonton, Canada, all have acquired **Aphex** Aural Exciters to sweeten sound for audio and video productions, announced **Marvin Caesar,** president of Aphex Systems Ltd. ITV Ltd. is utilizing its Aural Exciter for sound mixdown for **MCA/Philips** DiscoVision videodisks. In addition, **ABC-TV** is using an Aphex Aural Exciter in producing the **"Lawrence Welk Show"** for syndication. Aphex Aural Exciters are also being used in recording studios and by concert artists and rock singers on tour. Artists using the sound enhancer in concert or on albums have included **Willie Nelson, Linda Ronstadt, Diana Ross, the Cars,** the **Bee Gees, Fleetwood Mac, Elton John, James Taylor, Wings, Donna Summer, Neil Diamond, and Barbra Streisand.**

• **Ian Robertson,** formerly engineer for **KING AM/FM,** one of Seattle's leading radio stations, is joining **Audio & Design Recording,** the UK-based Signal Processing equipment manufacturer.

• **Neve Electronic Holdings Limited** of Cambridge, England has announced the appointment of **Clary A. MacDonald** as President of **Rupert Neve** of Canada Ltd. MacDonald has been employed by Neve since March of 1974, first as Canadian technical services manager, and later as marketing manager and vice president and general manager for Neve's Canadian operations. Prior to joining Neve in 1974, MacDonald was in the service of the **Canadian Broadcasting Corporation** as broadcast engineer.


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