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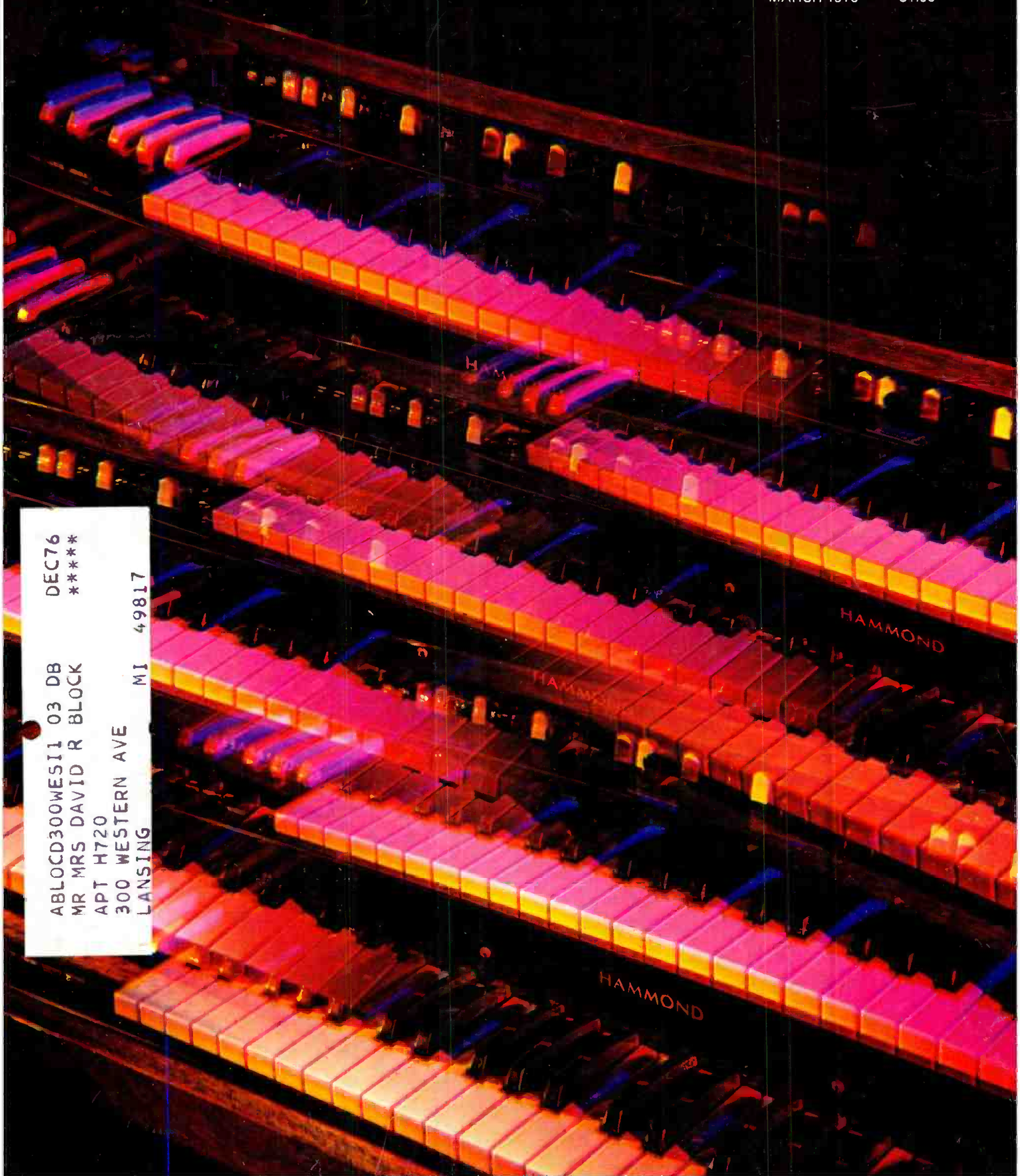
THE SOUND ENGINEERING MAGAZINE

MARCH 1976 $1.00

IN THIS ISSUE:

- Frequency Shifters for Professionals
- Understanding Harmonic Distortion
- F.M. Stereo Separation



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db roams afield in April, considering remote recordings of live performances.

- Well-known lecturer Don Davis, with Ron Wickersham, discuss Experiments in Enhancement, the delicate process by which correctly placed amplification equipment enables artists, particularly those using synthesizer-computer instruments, to control creatively the enhancement of their musical interpretations.

- R. A. Neilson and Bobby Goldstein report on an achievement by the Wally Heider people, recording live a combined *Beach Boys* and *Chicago* concert under terrific pressure of time and complication. The special ingredient of the professional, along with expertise, is pinpointed by the authors as Zen and the Art of Recording.

- Shifting to the broadcast scene, Patrick S. Finnegan discusses Dolby B and F.M. in his column. Add to this our other regular columnists, Norman Crowhurst, Martin Dickstein, and John Woram.

about the cover

- An unusual combination of creative lighting and experimental photography has produced this colorful montage of a Hammond organ keyboard. *(Credit: H. Armstrong Roberts)*

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THE SOUND ENGINEERING MAGAZINE

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db is listed in Current Contents: Engineering and Technology

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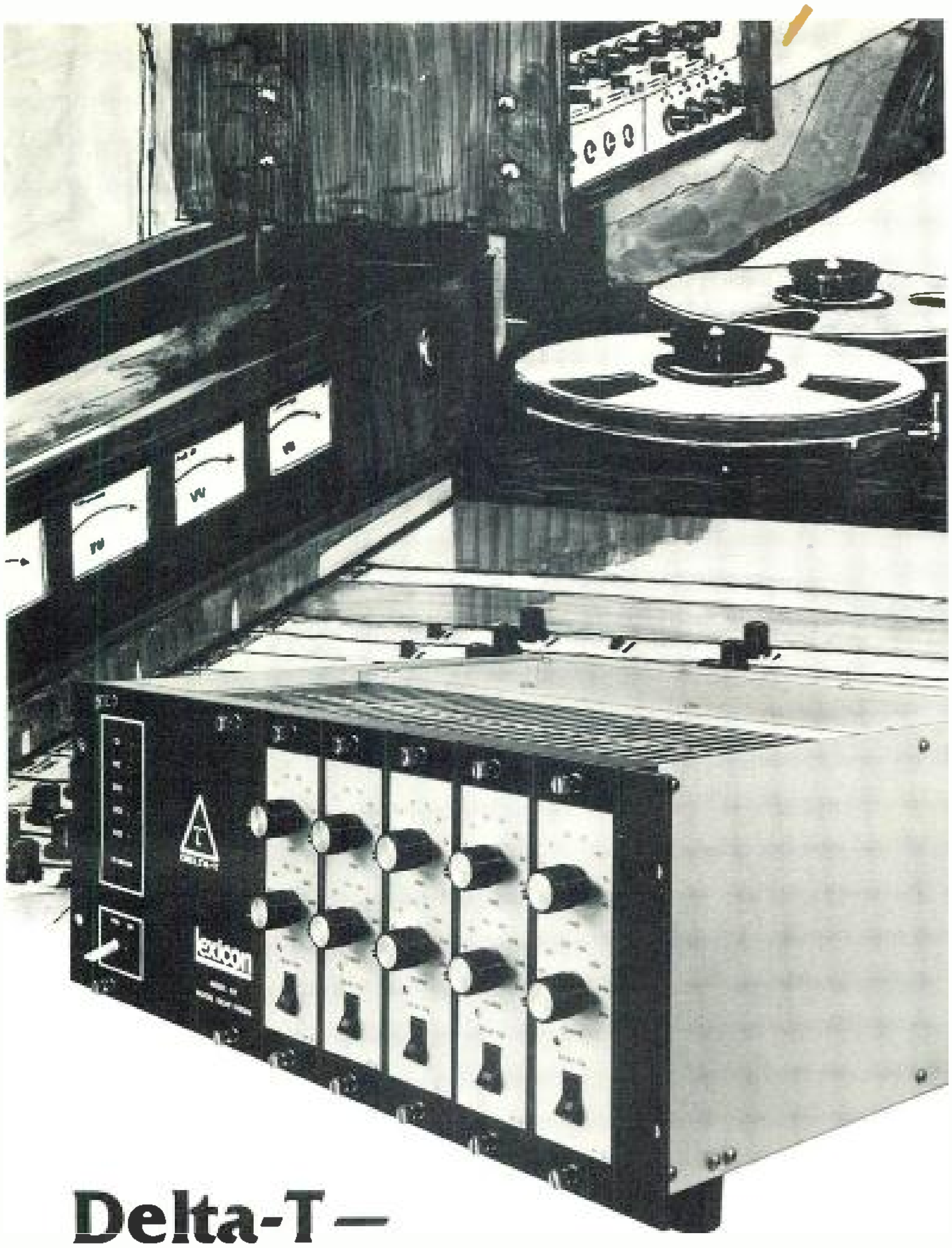

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db letters

THE EDITOR:

I have at least six Scotch metal 10½ inch reels that defy any attempts to de-warp them. They are valuable reels, but right now they scrape against the stainless steel panel on my Revox A77. Would any of your readers have comments? Thanks.

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CALENDAR

MARCH

21-24 **National Association of Broadcasters Convention.** Chicago, Illinois. Contact: NAB, 1771 N St., N.W., Washington, D.C. 20036. (202) 293-3500.

29-31 **NOISEXPO '76,** Hilton Hotel, New York City. Noise and vibration control. Contact: NOISEXPO, 27101 E. Oviatt Rd., Bay Village, Ohio 44140. (216) 835-0101.

APRIL

5-9 **Acoustical Society of America.** Washington, D.C.

22 **Acoustical Conference.** Hungarian Society for Optics, Acoustics, and Cenematography. Budapest, Hungary.

26-27 **Acoustical Problems of Light-Structure Construction of Buildings.** Acoustical Commission of the Hungarian Academy of Sciences. Budapest, Hungary.

MAY

1 **Midwest Acoustics Conference.** One-day meeting covering signal processing and data reduction technology for solving technical and legal problems in acoustics. Norris Center, Northwestern University, Evanston, Ill. Contact: H. O. Saunders, Rm. 24A, 225 W. Randolph St., Chicago. Ill. 60606. (312) 727-4331.

4-7 **Audio Engineering Society Convention,** Hilton Hotel, Los Angeles, Ca. Contact: A.E.S., 60 E. 42nd St., New York, N.Y. 10017.

28-31 **Sound and Vision '76.** Birmingham, England.

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db theory & practice

NORMAN H. CROWHURST

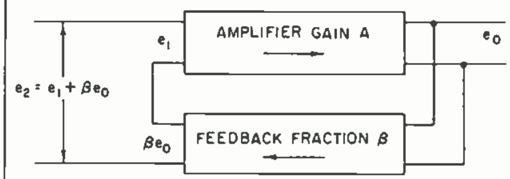

Figure 1. The classic feedback block schematic: voltage in, voltage out.

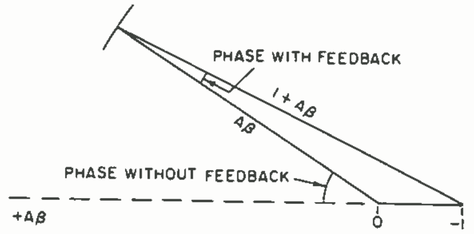

Figure 6. The construction for the Nyquist diagram for criterion of stability.

● Recently, I received a letter questioning my use of the word "mediate" and asking about Nyquist diagrams, referring to my article on Feedback in the November issue. The better known meaning for "mediate" is to act as an intermediary. We grew up with that meaning. But educators are always coining words. They use the word mediate to mean "put into media form."

Thus, when a lesson presently printed in a book is dictated onto tape, for example, it is being "mediated," in educational parlance.

Regarding the Nyquist diagrams, the reader admits that he should look back at the Part 1 of the series, because he only has difficulty with part 2. It is so difficult to keep from bringing up what I've said before. In Part 2, I started from the formula developed in Part 1 (September issue). And back last year, I commented on the little interchange about the use of formulas that occurred during the summer session at Brigham Young University.

What those students wanted was all the relevant formulas, so they could "plug them in" in the appropriate places for the audio systems on which they worked. My response, in brief, was to indicate that they need, far more, to understand what they are doing. Now this reader's query about Nyquist just confirms what I was saying there, once more.

In part 1 of the series, I gave schematic diagrams of feedback amplifiers, using series and shunt derivation of the feedback signal, at the output, and also series and shunt injection of the feedback signal, at the input. From this, in each instance, I derived the fomula that showed the effect of feedback on amplifier gain.

Also, in Part 1, I deliberately stayed away from phase angles, assuming just for simplicity, that feedback is always either positive or negative, never "in between." Of course, the facts of life are that there is never a feedback system that does not have in-between conditions as well. And that is what Nyquist diagrams are all about. But I kept that for Part 2.

As I had discussed the formula pretty well in Part 1, I felt that Part 2 could assume that Part 1 had been read and apply the formula to cases where feedback is not just positive or negative, but where it goes in between. In the series, I used more or less conventional symbols, mainly for the benefit of people who may have learned the subject before, but never understood it. But I am well aware of the difficulty of relying on formulas to convey a picture of what is happening. Vectors run into a similar problem, mainly because of the poor way they are too often taught.

Let us take a look at FIGURE 1, reproduced from the September issue. If you don't like all those symbols, disregard everything except the input end, for the moment, where you have three voltages: that at the input to the amplifier, labeled e_1; that coming from the feedback network, labeled βe_0; and that across the combination, which is what must be applied as input to the whole system, labeled e_2.

The equation following e_2 merely says, in algebraic terms, that these three voltages must jibe. Thus, if the internal input to the amplifier is 1 mV, the fed back signal is 9 mV, and the feedback is negative, the external input to the amplifier, e_2, must be $9 + 1 = 10$ mV. There is no way it could come to something different.

That could apply to d.c., or to a.c. of some frequency, which in general it more often does, producing what we call *signal*. We can think of d.c. and of signals of different frequencies, one at a time, but that simple formula must always be true. The voltages must add up round the 3-sided loop, at the input to the amplifier.

POSITIVE FEEDBACK

That was for the case of negative feedback. What about positive feed-

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back? Well, that can cause oscillation, which is why we need, later on, to get into Nyquist plots. But first take the simple instance, where we know feedback is positive, instead of negative. If the amplifier input is still 1 mV, internally, and the feedback is also 1 mV, but positive instead of negative, then we do not need any external volts at all for the amplifier to oscillate. If e_2 is zero, the feedback will provide the input voltage directly, and signal will go on forever —oscillating.

If feedback is positive, but less than equal to the original input, gain is increased instead of reduced. Thus, if now the *internal* input voltage is 10 mV, and the fed back signal is 5 mV, positive feedback, then the external input voltage needs only to provide the other 5 mV to make up the 10 total. Gain has doubled, because it takes 5 mV to produce the same output that 10 mV did without feedback.

If feedback, using the same example, is 8 mV, then gain is 5 times, because now it takes only 2 mV to produce the same output. If feedback is 2 mV, then gain is increased by 25 per cent, because we get the same output for 8 mV input instead of 10 mV.

PHASE

We are still with Part 1, conveniently ignoring phase. But phase won't go away because we choose to ignore it. As Part 2 started out explaining, whenever you use a capacitor or inductor, or something that has those properties, there are always phase shifts waiting to come out at some frequency or other.

There is no way to get a roll-off without phase shift, although there are ways to get phase shifts without roll-offs, which is another whole story. If you will now look at FIGURE 6, which was in Part 2, you can see how the same idea, easily accommodated by simple addition or subtraction when feedback is conveniently either pure positive or pure negative, can be applied to other phase combinations.

Now, the internal input voltage is that shortest line between 0 and −1. The feedback voltage is the line labeled $A\beta$. And the third side of that triangle, labeled $1 + A\beta$, is the external input voltage. These three must jibe by forming a closed triangle, because we have those three points in the circuit.

FIGURE 6 assumes that all the phase shift is in the amplifier, none in the feedback. So as well as being the fed back signal, the line $A\beta$ will have the same phase as the output voltage. Without feedback, the input voltage is the line between 0 and −1. But with feedback, it becomes the line labeled $1 + A\beta$. So this diagram enables us to show the effect of feedback on phase as well.

When we considered the simple positive or negative feedback cases, we showed that if the feedback signal is equal to, or greater than, the input signal, and positive in phase, the amplifier will oscillate without benefit of any external signal. In terms of the diagram at FIGURE 6, this means that the line labeled $A\beta$ will swing round to right until it passes through the −1 point to extending beyond it.

The Nyquist plot is made by constructing many of these diagrams, one for every possible frequency, and then joining up all the points where the apex of the triangle comes. The curve so formed is what mathematicians call the *locus* of the point, which merely means it is a curve showing how the point moves, as fre-

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quency, which is the *independent variable,* is changed.

Now a question some ask is, "Will the amplifier oscillate, if the frequency at which oscillation occurs is not present?" The answer is yes. Why? You've heard of noise, unundoubtedly. No electronic device is without it. If well-designed, noise is very low, hopefully inaudible, but still there. And noise contains a random sampling of all frequencies.

So, if at some frequency the line $A\beta$ extends through the -1 point, which means the curve its locus makes will encircle the -1 point, then at that frequency the random piece of noise will "go around again" amplified to a bigger level. Each time around will make the signal bigger at that frequency, until the amplifier is in full fledged oscillation, limited by distortion that puts components of other frequencies into the signal.

I hope that this additional explanation will help any readers who may, like the one who wrote in, have had difficulty with Nyquist. In a teaching situation it would be much easier. If instructional material is (what educators call) mediated, it can be made easier than it is, the way we are now doing it.

When I wrote that three-part series, although I tried to make it easy to follow, I was limited to using conventional written communication. The reader who wrote in asked a question that could have been raised immediately, had we been in class. Now—several months later—I respond to that question. Have I now made it clear? It will be months again before we know that.

That is an advantage of using media in education, when it is properly used. Responses can be built into the system. This kind of difficulty can be anticipated, and something put in to start the student on finding his way out of it. The point I had been trying to make is that when material is intelligently "mediated" this can be done.

In fact, since books are cheaper than mediated materials I see no advantage in mediating, unless the mediated material does something the books could not do. Merely dictating books onto tape is not, in my opinion, "mediating." For this reason, audio people have a responsibility to do something about education. At least, that is the way I see it. ■

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CORRECTION

In Martin Dickstein's January column, p. 18, line 6, "6328 Angstroms" was incorrectly referred to as "6,328 degrees." The correct terminology is Å. Angstroms are linear measurements, and canot be expressed as degrees.

db the sync track

JOHN M. WORAM

• Not too long ago, the New York section of the Audio Engineering Society had a meeting on the advance of technology. The panelists held forth on the state of the art today, as compared to the earliest days of commercial stereo.

The meeting came about as a result of some frequently heard complaints about the sorry state of some of today's recordings. Of course, there were a lot of wretched recordings released in the early days—most have long since achieved the oblivion they deserved. The good ones linger on though, and people sometimes ask why, after almost a quarter of a century of progress, these "golden oldies" still stand up so well, especially in comparison to some very recent releases. Or, if some 1950's records are so great, why aren't more of the hits of the 70's at least comparable, if not better, in overall recorded quality?

With the technology available today, we have the capability of producing great sound. But we also have the capability of thoroughly botching up a record. To prevent this, the mixer must be even more of a technologist than before, and yet he cannot forget musical values either. However, many of today's records subvert the music to the technology, as panelist Bert Whyte pointed out at the meeting. He happened to be talking about some recent classical quadriphonic recordings, but the remark applies as well to the top 100 scene.

Often, the technology gets misused because the engineer doesn't have the background that his craft really should demand. Or, his producer is, to put it bluntly, an incompetent.

At the meeting, someone referred to the wealth of information available in any well equipped technical library. Someone else pointed out that much of that information is written in "Technicalese" and can only be comprehended by people with advanced degrees, or a talent for the obscure. Faced with this mountain of difficult reading, the beginner is apt to throw up his hands in despair at ever finding a paper he can understand.

Two things are needed. The first is to dispel the notion that in order to be a successful engineer, all you need to know is how to snap your fingers on the beat, and when to say "outasight" or whatever they say in your town. The second is some sort of guidebook through the academic jungle for those who really want to learn a little something about recording.

Which brings us more or less to the point of this little epic. In working on my book (subliminal plug) I've managed to accumulate a small collection of papers on this and that subject, one of which is plain old stereo. Some are more readable than others, and most have at least a little something of interest to the working recording engineer. Stereo may be old, but it's not so plain, and you don't really get it from a bunch of pan pots. contrary to popular belief. It seems there's a lot more going on out there in papersville than many mixers dream of, and some of it may even help you get a little more out of your recordings.

GUIDED TOUR THROUGH THE JUNGLE

So-o-o, this is sort of a guided tour through at least a few of the papers that may be of particular interest. Authors have been writing on the subject for years, and much of it is relevant today, especially with multi-track technology. We can start off with a little quiz.

1. How many ears do you have?
   a. 1
   b. 2
   c. 3 or more

The correct answer is b. If you answered a or c, you're a special case. and this article is not for you.

2. If you have only two ears, what are you doing with all that multi-track recording gear? (Essay type answer on this one)

"I'm using it to create all sorts of beautiful music which would otherwise be impossible."

3. When you get all finished creating all sorts of beautiful music which would otherwise be impossible, how many ears will you have?

3. a. 1
   b. 2
   c. 3 or more

The answer to this one is also b.

EARS

So, no matter what you do or how you do it, it all comes back to two ears. Last month's **db** article by Dan Queen had a little something to say about how the ear works. Not just his ear, but yours too.

For instance, let's say we're doing a mixing session, and want the guitar on track 15 to be right-of-center. The pan pot should take care of that nicely. But before you reach for it, think about what you would hear if the guitar was not on the tape, but in the room with you, sitting just to the right of your center line.

Common sense tells you that you would hear the music with both ears, and the intensity difference from one ear to the other would probably be unmeasurable. Yet you would know exactly where the guitar was located. Why?

More than twenty years ago, William B. Snow wrote: *A complex wave pulse has an initial wavefront which arrives at the near ear a short time before it arrives at the far ear. It is this small time difference which is used by the hearing sense to determine small angular variations, particularly for sounds near the median plane (straight ahead) . . . The loudness differences at such small angles are negligible and it must be assumed that the arrival-time differences give the localization clues.*[1]

Note that Snow emphasizes the importance of time of arrival, rather than intensity.

Now then, back to the pan pots.

You've placed the guitar right-of-center by feeding track 15 to both speakers, but in unequal proportion. The sounds from the speakers arrive at your listening position at precisely the same time, and of course both ears hear both speakers. Each speaker tries to convince you that the sound is coming from it alone. As long as you remain well centered, your hearing mechanism doesn't have much trouble refereeing this psycho-acoustical tug of war, and the localization is reasonably effective

But if you move around, the guitar moves with you. In fact, if you move to that right-of-center location, the sound from the left speaker is now delayed slightly—just the opposite of what would happen if the guitarist was actually in the room, and you moved to a seat directly in front of him And, as you move closer to one speaker, it (apparently) gets louder. In a review of the Haas Effect, Mark Gardner describes what happens: *If now, one of the real sources is slowly moved farther away, the apparent source moves towards the other (nearer or earlier) signal. If one signal is made stronger than the other, a similar movement will occur toward the louder signal, or an interchange between level and time of arrival can be made within certain limits.*[2]

Bringing all this back to the world of recording, it seems as though the pan pot is not the greatest directional tool in the world. As Gardner implies, it has "certain limits." But, when mixing down a multi-track tape, it may be all you've got at hand. even though in 1958, Fr. Heegaard wrote: *It has generally been considered that, in stereophonics, it is preferable to rely on a single pair of microphones in order not to spoil the directional effect.*[3]

Needless to say, this policy would severely cramp the style of a lot of contemporary recordings, but it's interesting to note that long before the birth of multi-track recording, some of its limitations were anticipated, at least in the literature.

So where's the happy ending to all this? Maybe the literature also suggests a way to make better recordings, as well as telling us what's wrong with the ones we're making.

TWO-MICROPHONE PICKUP

Well, almost. There are many references to the excellent sense of stereo perspective when one or another type of two microphone pick-up is used. Carl Ceoen compared six different microphone placements (five stereo and one pan pot) and in most of his tests, the pan pot method was outranked by one or more of the stereo placements.[4] Earlier, an application note from Gotham Audio Corp.[5] described a method of mixing additional several stereo pairs together. The technique is quite interesting, but needs a separate article to describe it fully. In practical terms, it has the disadvantage of requiring two tracks for each additional stereo mic if the engineer is not prepared to mix them together during the recording session.

Since this practice is an unlikely one (especially on Sel-Sync sessions!), it may not be of much help to the modern we'll-fix-it-in-the-mix technician who is trying to come up with a better recording. But, what about when it comes time to add the soloist? Maybe a little extra effort would pay off here. Perhaps some of the stereo techniques that have gotten pushed aside should be dusted off and tried.

Can you spare two tracks for the soloist? Why not have him/her/it sing into a crossed pair of Figure-8s? If you've really got nerves of steel, have the chorus stand on the other side of the microphone and do their thing at the same time. If you can get the producer to *listen* before he has his coronary, he may actually like what he hears. Then you'll be ready for the real hard-core stuff, like miking a whole darn string section in stereo! Of course, the burden of musicianship is then passed back to the musicians, who may not be ready for such a shock. But if you explain it very carefully, they may actually get enthusiastic about playing real music again. Or they may walk out. It's happened before. ■

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db sound with images

MARTIN DICKSTEIN

• When we began this discussion last month, I mentioned some of the complex installations of audio-visual systems designed by Hubert Wilke Associates of New York City. Most of them include various projectors, such as the overhead and the film units, and also one or more slide projectors. Some are front-screen, others have rear projection. Usually large facilities also include remote controls so that the presenter can advance or reverse the slides, start the film, play a tape, etc. Some control units allow volume adjustment, light dimming, curtain movement, and random access among other environmental and program operations.

In addition to the projector, there are two other considerations that are also vital to a successful showing. One is the presenter himself; suggestions were made in the previous column about ways someone involved with selling equipment or designing a/v facilities might be of assistance to the client by offering tips to help the presenter make a better showing. The other factor is the software to be used. No matter how sophisticated the installation, and how polished the presenter, if the software falls flat, part of the message is bound to be lost, along with the expense of the material itself.

The production of films, or filmstrips, is an art in itself. Many companies have been formed for the purpose of producing films for specific applications. Some specialize in training material. Others produce travelogues, cartoons, commercials, or stock material for cutting into other films. These films can become quite expensive, depending on location, staff required, casting, length, editing and laboratory work, and so on. There is one item, however, that can be deadly boring if no originality is exercised in the production . . . slides.

SLIDES

Almost all presentations made with a projection system include slides. The usual routine is to show a slide with words on it. Most presenters like to read the words, then talk about the subject. Others show the slide but do not read it. They talk about the subject but leave the viewer to read for himself. This can prove to be very distracting for most of the audience since they don't know whether to read or listen, and they can't do both.

When there is only one projector in use for a single-image presentation, there is the usual 1½ second black space between slides. If a monotonous presentation, one slide after another, is followed in a regular sequence, it can become sleep provoking. For someone in the audience who just finished a heavy lunch with two or three drinks, it's like driving at night with heavy eyes and becoming mesmerized by the white dashes of the lane-dividing line flashing by. To prevent this, not only should the presenter be more animated to keep the audience's attention alert, but the equipment can be used to greater advantage, varying the length of time used for each slide to create greater interest and increased retentivity.

When two projectors are available for side-by-side showings, this can add to the impact of the presented material. They need not be advanced simultaneously. Provision can be made to work one unit while the other remains stationary. This, then, allows a change of slide on the left while the right side is black. A complimentary slide can then come up on the right, and change several times while the one on the left is stationary. Then both can go, and the right side come up alone.

It might also be an interesting arrangement, if a film is used with the slides, to have it displayed on one side, in place of one of the slides, instead of in the center. This way, the slide on the other side of the screen can mention the point under consideration while the film is playing. This leads to one more interesting possibility. The slide shown just before a film is to go on can actually be the first frame of the movie. This way, the film can overlap the slide for an instant before the slide is advanced to a black. It will look as though the slide had started to move from a still-frame of the film.

THREE PROJECTORS

Where a third slide projector is available, a variation that is possible is to have either three side-by-side slides, or, maybe even more effectively, an interlace of center screen images with two side-by-side slides, so the viewers' eyes have to vary their positioning toward the screen at different times. Sure, you can now add a second film projector for either a showing in the center or on the other side of the first film unit, but there is a limit. There is such a thing as overkill. If the effect to be presented is for mood, or to indicate complexity of a situation, then anything may go. If, however, a definite message is to be presented, with facts to be remembered, let it not go overboard. But by all means, keep the audience awake by using the equipment or installation to its best advantage.

HORIZONTAL OF VERTICAL SLIDES

Now that the equipment and the presenter are ready for the presentation, how about the slides themselves. In the simplest setup, the single image from one projector, the slide format is 2:3 (height to width). The horizontal format has several advantages. When you consider the usual room with a flat floor and a 7-to 8-foot ceiling, and a seated audience, the bottom of the image should not be lower than 4 feet off the floor. Possibly 3½ feet, but if it is lower than that, the people toward the back will have great difficulty seeing the lower part of the picture.

This allows a 3- to 4-foot high image. The width, correspondingly, would be 4½ to 6 feet. This would permit good visibility to a last row somewhere about 27 to 36 feet from the screen, with application of the rule-of-thumb, 6x image width. With proper letter size (another story for sometime in the near future), and good artwork, there should be no problem getting an effective message displayed on the screen. In order to keep the slides moving with interest, it's best to use good pictorial representations such as photographs, instead of words where possible, logos instead of names, shapes in place of straight-forward typed copy on the slide, and so on.

Since a good deal of the material usually read is vertical in shape, such as newspapers or magazines or books, or even advertisements, it sometimes is well to use vertical slides. However, in many presentation rooms, vertical slides would spread over the top and bottom of the horizontal-shaped screen and look bad. A vertical effect is possible, however, by shooting the vertical material in a horizontal format but on a black background. This avoids showing the shape of the slide not being used. The copy has the appearance of being vertical, and is a definite change from the other slides which may be horizontal.

If you use white words on a dark background, a harsh contrast is created, causing eye fatigue after a while. In some cases, where only a few words or a symbol or logo are used, they might be on black for impact and change from the slides around it. But, especially, where the system is front-projection, harsh contrast just doesn't work well over a prolonged presentation since the lights in the room are probably subdued, or close to dark. Even in rear-screen systems, continued viewing of sharp contrast, in spite of the fact that the lights can be left on in the room during the presentation, can be tiring. However, for effect, impact, variation, and movement, contrast can be of some value.

DISSOLVE SYSTEM

A simple variation on the one-image theme is a dissolve system. This eliminates the 1½ second black pause between views while the mechanism advances the slides. A smooth movement between slides can be effective in building a graph, for instance, or a chart, or a pictorial image. Starting with a single bar on a chart, dissolving into a two-bar chart, then three bars, etc., can be effective in showing growth. Dissolves from a small image to a larger one introduce movement and indicate growth. (In the case of the recent economy, perhaps the dissolves were shown in descending or receding order.)

One way to show detail on a complex chart is to dissolve into close-ups of the desired section of the original large-scale image. All sorts of variations can be designed with great effect and a show of creativity. Of course, in a dissolve presentation, remember that slides alternate from one projector to the other. This may cause some problems with changes in the presentation, especially if the changes include moving slides around and go up to the last minute, too. The presentation, however, will gain from variations in the material.

You, as the supplier, installer, designer, recording engineer, photographer, producer, user, or technician, can really be of great help to the client if you can show him how equipment and systems, slides and software, and the presenter himself can help to improve the presentation and make the message get across . . . and stick. It takes all three to tango. ■

PATRICK S. FINNEGAN

F. M. Stereo Separation

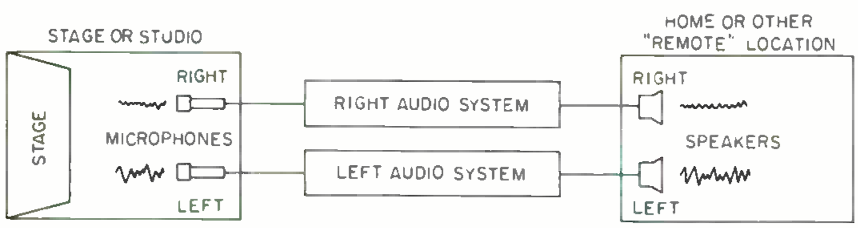

Figure 1. The basic purpose of stereo— a pair of "remote" ears.

A VERY IMPORTANT element of a stereo system is separation of left and right audio channels. Without separation, the two channels would blend into one and the system cease to be stereo. Maintaining separation is difficult enough in an ordinary audio system, but it is far easier than maintaining separation through an f.m. transmitting system.

SEPARATION

An individual listening to a live performance on stage hears sounds from many directions. Since he normally hears with two ears, he can discern a sense of direction from which the sounds come. The stereo system attempts to capture these sounds on at least two microphones and direct the output of each microphone through separate channels into storage on audio tape or record, later reproducing channels through two separate speakers. In other words, the system tries to provide "remote" ears for the listener.

Patrick S. Finnegan has had a long and distinguished career in broadcast sound. Beginning next month, Mr. Finnegan will contribute a regular column Broadcast Sound.

These two channels, then, must faithfully convey the original sounds as obtained by the microphones to a speaker output for each channel. If sounds are permitted to blend together haphazardly, any place along the route, the channel separation will be lost.

DETERIORATION

Many elements occur between the two microphones and the final two speakers. Each element has its own limiting factors which can deteriorate or destroy the channels' integrity. Besides the microphones, there are the various amplifiers, the recording tape machines or disc cutters, the reproducing machines and amplifiers. When it is desired to send the stereo through an f.m. system, a great many more elements and limiting factors are introduced into the chain.

An f.m. system does not transmit left and right audio channels separately, as is done through an audio system. Instead, the left and right audio channels are carefully blended together in a matrix system. After further processing, the sound comes out of the stereo generator as a *composite* signal. It is this composite signal which actually modulates the transmitter. Although the basic requirement of the stereo system is that the channels remain separate, the matrix deliberately blends these two channels together.

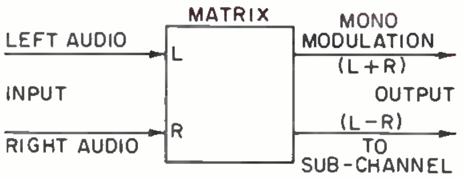

Figure 2. The matrix blends the two audio channels to produce SUM and DIFFERENCE signals.

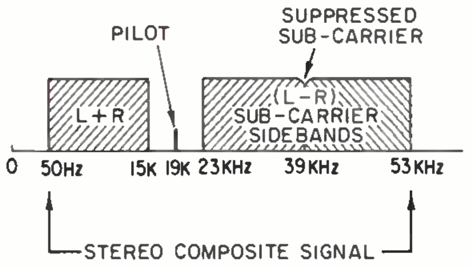

Figure 3. The composite signal modulates the transmitter. It is composed of many signal elements on into the supersonic region.

Herein lies its greatest potential for loss of channel integrity, for, if this is not done carefully, the channels cannot be recovered and restored to their full integrity in the receiver.

THE MATRIX SYSTEM

Any broadcast system that is well established in public use is required by the FCC to provide a compatible signal when any additional service or modification is done to the original service so that the public will be able to receive the same service on their existing equipment as they did before without degradation. It is for this reason that color television had to develop a signal compatible with black and white receivers; for the same reason, quad, or 4-channel stereo, is going through the same throes.

A matrix works in this manner. The output of the station's left and right audio channels terminates at the left and right input of the stereo generator, where the signals go directly to the matrix. The matrix adds the left and right channels to produce a *SUM* (L+R) signal. At the same time, another part of the matrix *inverts* the right channel and combines it with the left channel to produce a *difference* (L−R) signal. The sum signal provides the compatible monaural signal for mono sets. The difference signal will then amplitude modulate a 38 kHz subcarrier, producing double sidebands. The 38 kHz carrier itself is suppressed and only the sidebands remain.

A synchronous detector is required in the receiver to recover these sidebands; this must be phased with the original carrier. The basic oscillator is a crystal-controlled 19 kHz oscillator in the stereo generator. The second harmonic (38 kHz) of this oscillator is modulated as the sub-carrier. The 19 kHz signal itself is transmitted to synchronize the receiver detector.

The composite output of the stereo generator has a bandpass extending from 50 Hz on up into the supersonic regions. It is made up of the L + R signal in the audio band of 50 Hz−15 kHz, a 19 kHz pilot signal, and double sidebands of the suppressed 38 kHz carrier that extend from 23 kHz to 53 kHz. This is the signal which modulates the transmitter and it will be the signal that is detected in a *wideband* demodulator in the receiver. It must be further processed by a synchronous detector to recover the L−R signal, and along with the L+R signal sent into another matrix that will restore the original left and right audio channels.

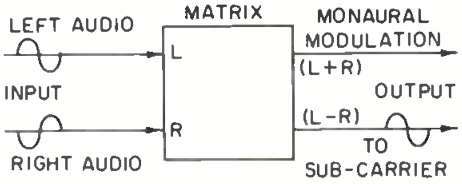

Figure 4. When audio input signals are 180 degrees out of phase, the matrix will shove all the audio into sub-channels.

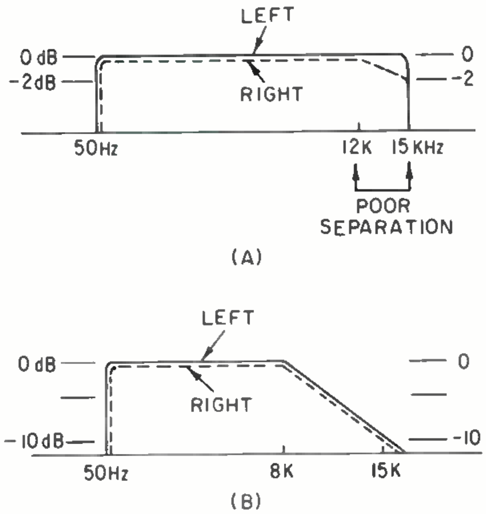

Figure 5. Response curves must be identical. In (A) both curves are good individually but not identical. Poor separation will result at 15 kHz. In (B) both curves are poor, but identical. The system will have good separation.

POOR SEPARATION

The matrix system relies very heavily upon phase relationships throughout. Anything, from the original microphones to the final destination in the receiver, which can distort the phase relationships can reduce the system's ability to recover and restore the original channels and their original integrity.

In the stereo generator, the sub-carrier and pilot must be properly phased, the transmitter working properly, with no high standing waves on the transmission line or antenna problems. Once all these have been originally adjusted properly, they generally remain stable, unless some component fails (such failures usually trip out circuit breakers or alarms and must be corrected). From an operational standpoint, the problems which most beset the stereo system are in the audio system itself. These occur in two general categories—phase and amplitude response of the left and right audio channels. When phase is wrong, the two signals do not reach the matrix at the same time or the polarity of one channel is reversed. And when the audio response curve of each channel is not *identical*, the varying response amplitudes do not permit complete cancellations in the matrix, so what remains shows up in the opposite channel and separation suffers.

Polarity reversal of one channel (180 degree phase shift) which places the two channels out of phase will cause the input signals to be shoved into the sub-channel and little on the main channel. This can be demonstrated by feeding a sine wave tone out of phase to both inputs of the stereo generator. With a sine wave, complete matrix action takes place and there is *no signal* on the main channel; it is all in the sub-channel. With program, the mono receiver would suffer a severe drop in signal level.

Assuming the original installation was correctly phased, reversal of polarity usually happens because a patch plug has been turned over, or the wiring has been put back on incorrectly during maintenance, such as replacing the head on a tape machine.

Phase shifts of less than 180 degrees cause a lead or lag between the audio channels' phase relationships. A major cause of this is the path length of each channel. Signals starting at the microphone together should reach the matrix input at the same time. If they do not, complete matrix action cannot take place, so the unprocessed part of one signal will show up in the other channel and separation will deteriorate.

Anything which can cause one channel to lead or lag behind the other channel will change the correct phase relationship. This can be caused by faulty or defective components in the audio system, but it can also be due to the original installation, where wiring lengths were not given careful consideration. All these differences are cumulative and a fixed, but incorrect, phase relationship is set up. There can also be other path length problems with Telephone Company lines to the transmitter site or when stereo remotes are done over Telephone Company lines.

AMPLITUDE RESPONSE

The response curve of each channel must be identical with the other, or proper matrixing cannot be done, and what is left uncancelled will show up in the other channel. Separation does not essentially have a relationship to fidelity, but instead, to identical response curves, whether these are good or poor, assuming there is no phase shift also involved. For example, each channel has a reasonably good response curve, but not identical to the other. One is flat all the way to

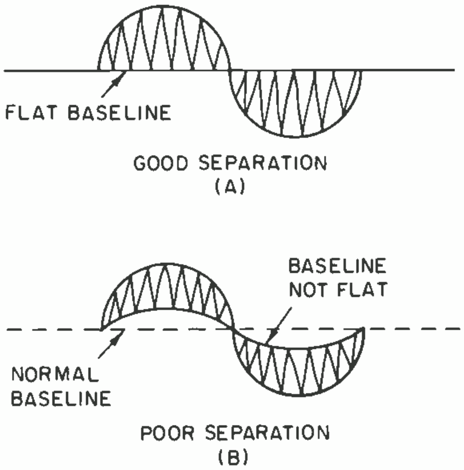

Figure 6. Use an oscilloscope to measure the composite signal out of a stereo generator.

15 kHz. The other one is flat out to 12 kHz but then rolls off 2 dB at 15 kHz. Both curves are in specs as individual curves go, but there will be poorer separation above 12 kHz. On the other hand, two identical but poor response curves, for instance, flat to 8 kHz and then rolling equally so the response at 15 kHz is down 10 dB, will have good separation.

Many, many faults or misoperations along the way can effect the response curve of the channels. There may be improper alignment of a tape machine head, a defective stylus in a turntable, improper level settings of amplifiers, impedance matching problems, misadjusted equalizers etc. Anything which can effect system response, *unless it effects both channels equally*, will show up as poor separation.

MEASUREMENT

We can listen to the signal off the air with a good receiver and obtain a qualitative measurement of the separation, but this does not tell us how much separation is present. To measure this, a sine wave generator and the modulation monitor can produce the information. Use the method described in the instruction manual for the monitor, but feed the signal to the *input* of the *audio* system to get the real separation figure. All this assumes that the monitor is properly adjusted and its own separation and phasing are correct. If the monitor is incorrect and the system adjusted to read correctly on the monitor, the system would be actually misadjusted, even though it would appear correct on the monitor.

To verify the monitor figures, feed a sine wave to the input of either the left or right channel and measure the output of the stereo generator composite signal with an oscilloscope. The base line on the scope figure should be flat or nearly so. Next, check the output of the detector in the monitor (the composite signal) and note the flatness of the base line. This check will measure the signal after it has passed through the transmitter. Assuming that the pilot phasing was correct, if there is not a very flat base line, tweak up the stereo generator adjustments. If this flattens the base line out at the output of the stereo generator but not much out of the monitor. there are some transmitter problems. But if it doesn't flatten out the base line after the stereo generator, then there are some audio system problems. In most cases, this is where the problem will be. So, you will have to go to work on the audio system, but look first for audio response problems. ■

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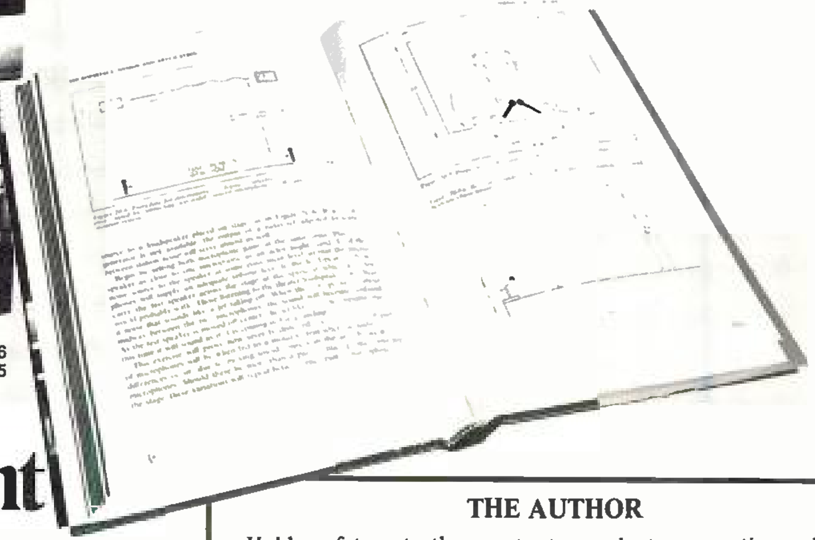

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This text is highly recommended as a teaching tool and reference for all those in the audio industry. *Price: $20.00*

THE AUTHOR

Holder of twenty-three patents on electro-acoustic products, Lou Burroughs has been responsible for extensive contributions in the development of the microphone. During World War II, he developed the first noise cancelling (differential) microphone, known as the model T-45. Used by the Army Signal Corps, this achievement was cited by the Secretary of War. Burroughs was the creator of *acoustalloy*, a non-metallic sheet from which dynamic diaphragms are molded. This material made it possible to produce the first wide-range uniform-response dynamic microphone. Burroughs participated in the design and development of a number of the microphones which have made modern broadcasting possible – the first one-inch diameter wide-range dynamic for tv use; the first lavalier; the first cardiline microphone (which ultimately won a Motion Picture Academy award) and the first variable-D dynamic cardioid microphone. He also developed the first wind screens to use polyester foam. Burroughs was one of the two original founders of Electro-Voice, Inc. He is a charter member of the Society of Broadcast Engineers and a Fellow member of the Audio Engineering Society.

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ELECTRONIC CROSSOVER

• Model X0312 keeps all outputs in phase at all frequencies and provides continuously tuneable crossover frequencies from about 100 to 1,000 Hz and 1,000 to 14,000 Hz. The manufacturer claims distortion of less than 0.1 per cent and a signal-to-noise ratio of better than 80 dB. High- and low-pass filters in each crossover are permanently crossed at 3 dB down. The state-variable filter gives a 12 dB per octave Butterworth response. The sum signal is flat ± 1 dB from 20 to 20,000 Hz. The crossover has balanced 600-ohm transformer outputs individually adjustable to a maximum of 8 volts. Voltage gain is adjustable to a maximum of 2.

Mfr: Stevenson (Interface Electronics)
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OSHA MEASUREMENT SET

• Sound level measuring set model 1983 meets ANSI S1.4 1971 Type S2A standards. The unit spans a single range, 70 to 120 dBA. Operation of the meter is quite simple, with no range or weighting selections to be made. The meter spans the 50 dB range in clearly marked 1 dB increments; OSHA limits are printed on the meter face. In addition to the meter, the set includes a sound-level calibrator, windscreen, carrying case, etc.

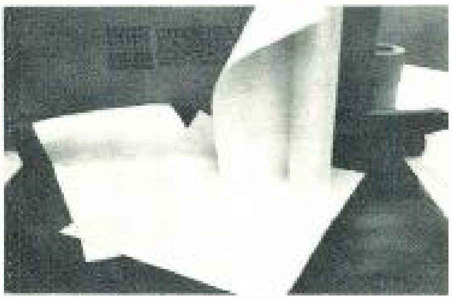
Mfr: GenRad, Inc.
Circle 51 on Reader Service Card

FOUR-WAY LOUDSPEAKER

• High sound pressure levels with moderate power input was aimed at in the design of model 7 loudspeaker. Particular emphasis is placed on transmission of upper bass and lower mid-range tones with fidelity. The loudspeaker employs six drivers. It has a filter network rather than a conventional crossover, which the manufacturer claims offers improved transient response and greater transparency. The speaker has been designed to handle the special needs of rock music, as well as other musical forms.

Mfr: Rectilinear Research Corp.
Price: $399.
Circle 52 on Reader Service Card

SOUND ABSORBING FABRIC



• Sheerfill 3A soundproof fabric is lightweight, flexible, and translucent. It can be easily cut and draped. The coated beige surface is easily washable and fire-resistant. Acoustical properties test out to NRC 0.60 minimum, ASTM-C423-66. The manufacturer claims a tensile strength of 200 minimum lbs./in. A thin fabric, Sheerfill combines with air in front and in back of the installation to furnish sound absorption.

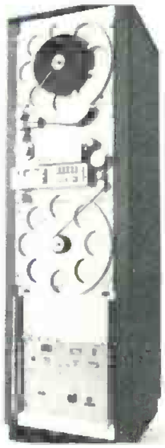
Mfr: Chemical Fabrics Corp.
Circle 53 on Reader Service Card

LIVE PERFORMANCE MIXER

• This fifteen-microphone channel mixer provides treatments for left and right p.a. and stage monitor with two auxiliary outputs. Each input channel has tone controls for bass lift or cut, treble lift or cut, and middle lift; a fourth tone control, continuously variable, governs the frequency of maximum/minimum lift. There is a continuously variable sensitivity control, switched input attenuator and a peak reading meter. Rotary or linear faders control monitor output and treatment channels. The device features independent two-track tape recorder level controls, listen and talk facilities, and two auxiliary peak reading meters. Five input channels are built as a single module and the output and auxiliary channels are built as another module of the same size.

Mfr: RSE (Lamb Laboratories)
Circle 54 on Reader Service Card

16MM TAPE RECORDER



● A tape drive system using a smooth capstan, claimed to give very short start-times, speed stability, low wow and flutter and reduced head wear, is featured in DS-16 16mm. perforated tape recorder/reproducer. Synchronization is fully electronic. Built-in memory circuits give full programming capability, tape search and all operational features to meet current and proposed recording standards. The unit is suitable for film dubbing as well as for straight record/replay.

Mfr: Schlumberger Instruments

Circle 55 on Reader Service Card

ECONOMY SPEAKER SYSTEM

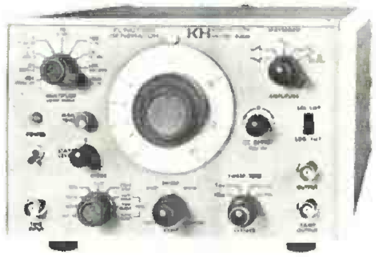
● Techniques for handling crossovers and in balancing phase lags and leads of multiple drivers, refined in the designing of this manufacturer's high-priced speakers have been used in the creation of economy 3-way Monitor Jr. All drivers (12-in. transmission-line woofer, 1½ in. dome midrange, 1-in. dome tweeter) deliver temporal information precisely in phase. The manufacturer claims a dimensional quality to the sound, reproducing orchestral sounds in relation to the instruments' position—left, right, front, or rear. Available in bookshelf or pedestal models.

Mfr: Infinity Systems, Inc.

Price: $225.

Circle 56 on Reader Service Card

FUNCTION GENERATOR



● Nine modes of operation are possible with model 5300 function generator. In addition to separate waveforms and ramp outputs, pulse, sweep, and burst modes, it offers an exponential ramp function for logarithmic sweeping. The exponential sweep in conjunction with the linear sawtooth output enables semilog plotting. There is external voltage control of main output frequency. Pulses can be as narrow as 200 ns at rep-rates anywhere between 100 kHz and 0.1 Hz. An adjustable trigger gives a one shot performance of either a single cycle of a waveform or a single frequency sweep.

Mfr: Krohn-Hite Corp.

Price: $695.

Circle 57 on Reader Service Card

MARC SAUL

Understanding Harmonic Distortion

Harmonic distortion generated by audio equipment discolors musical output, sometimes objectionably. Testing and making the necessary adjustments will keep the distortion to a minimum.

HARMONICS, which are multiples of a fundamental frequency, are what give music and speech its particular character and timbre. Without harmonics, music and speech would sound dull and lifeless and it would be difficult to distinguish one voice or musical instrument from another.

For example, assume we strike the low-C note on a concert grand piano. In addition to the fundamental sine-wave frequency of 32.7 Hz being produced, harmonics of up to about the fiftieth of the fundamental tone will be generated. Furthermore, since the piano sound board is not large enough to radiate frequencies much below 50 or 60 Hz, the fundamental may be missing altogether. The output waveform then consists almost wholly of the harmonics (see FIGURE 1). Without these harmonics, and particularly the higher order harmonics, the note would practically disappear or would sound muffled and without a distinct piano character.

Audio equipment, such as line or monitor amplifiers, recording or playback amplifiers, tuners or receivers, and cutting heads or loudspeakers, are not musical instruments. This equipment must not introduce harmonics of their own so that they color the tone of the instruments or voices being handled. Instead they must be nearly perfectly transparent as possible to the sound signals being amplified or being transduced. By the extent that they introduce their own harmonics or other signals, they produce distortion.

Since no piece of audio equipment is perfectly distortionless, some distortion must be tolerated. The idea, though, is to have as little distortion as possible so that the audio signals being handled are as free from alteration as possible. With good equipment, the amount of distortion will be below the level at which it can be perceived.

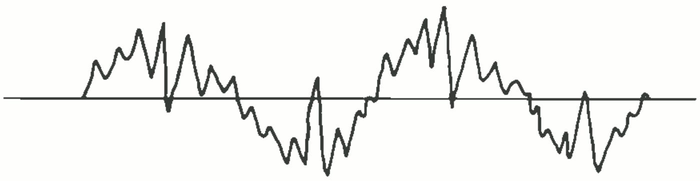


Figure 1. Waveform of a low note struck on a piano.

HARMONICS AND NON-LINEAR DISTORTION

Audio equipment is subject to several different types of distortion. But one of the most important is harmonic distortion. This occurs when the equipment being used changes the waveform of the signal being handled in the same way that it would be changed if harmonic frequencies were added to it. It also occurs when the equipment alters the size and shape of the harmonics already in the signal by either boosting or attenuating these harmonics.

Now consider a perfectly pure 400-Hz sine-wave signal being applied to a recording amplifier. If the amplifier has no harmonic distortion, then the output waveform would be a magnified but otherwise wholly unchanged replica of the input (FIGURE 2A). However, if the outputs are as shown by the solid-line waveforms of FIGURES 2B through 2H, then harmonic distortion is present. The amplifier acts as though it were adding additional fre-

Marc Saul has for many years been a writer and editor, covering the audio scene.

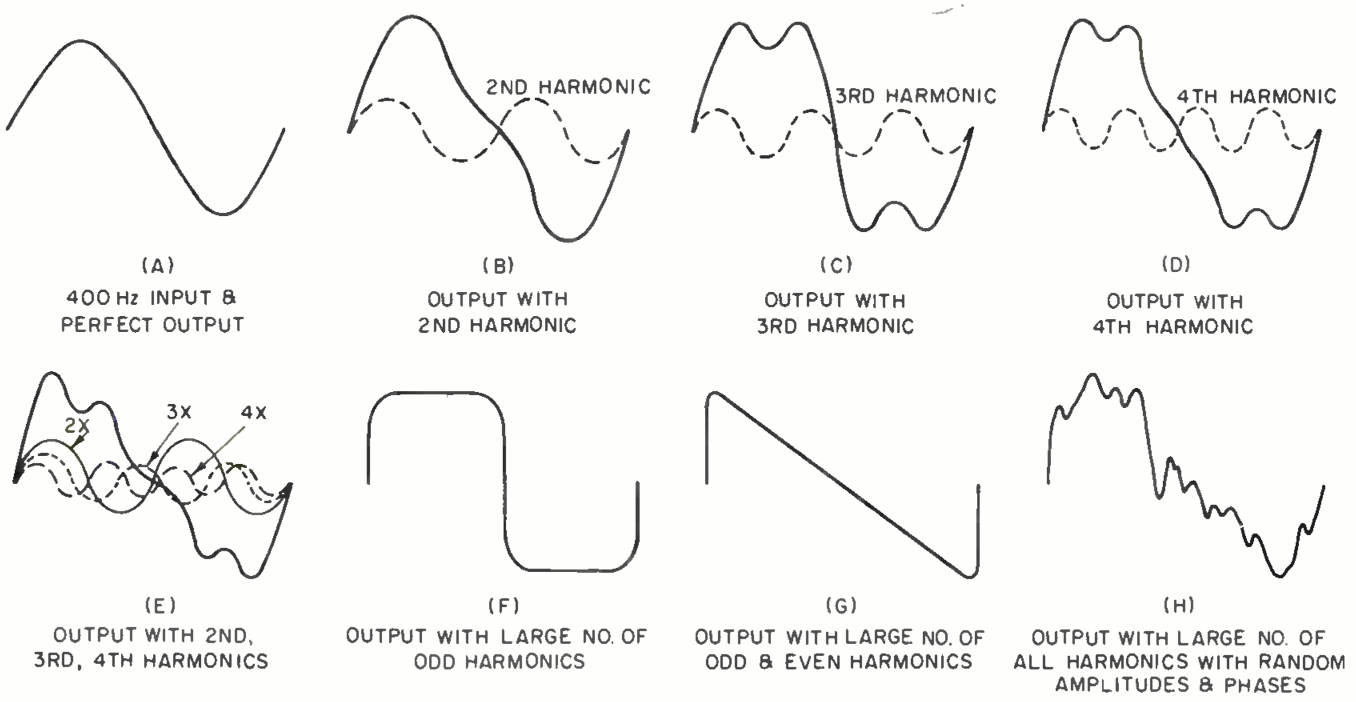


Figure 2. Waveforms with various numbers of harmonics.

quencies which, when added to the fundamental, result in the distorted waveforms shown.

With second-harmonic distortion, the sine wave takes on a skewed appearance and the downward slope has a couple of ripples in it. With third-harmonic distortion, the peaks of the waveform are changed into dips. When fourth-harmonic distortion is present, both dips and ripples appear. In a case where there are a very large number of odd, in-phase harmonics the waveform takes on a square appearance (FIGURE 2F). Hence, if the amplifier clips both positive and negative peaks of an incoming sine wave, the effect is as though a large number of *odd* harmonics have been added. With a large number of *all* harmonics, the waveform is converted into a sawtooth (FIGURE 2G). Finally, with numerous harmonics having random amplitudes and phases with respect to the input, the irregular output waveform in FIGURE 2H would result.

Next, consider the input-output linearity on the transfer characteristic of a piece of audio equipment, such as a playback amplifier. If the amplifier were perfectly linear, its transfer characteristic would be a straight line as shown in FIGURE 3A. With a sinusoidal input voltage, the output voltage would be a replica of the input—also sinusoidal.

If the amplifier has a transfer characteristic that is nonlinear, as shown in FIGURE 3B, the output would have a positive peak that is flattened out while the negative alternation is normal. The result is a disorted output waveform. A waveform such as this, with positive and negative half cycles of different shapes and areas along with a steady (rectified) d.c. component, which in this case is negative, has *even*-harmonic distortion.

With a different type of nonlinearity, as shown in FIGURE 3C, the S-shaped transfer characteristic results in an output waveform that is tall and peaky. The result again is distortion. A waveform such as this with positive and negative half cycles similar in shape has *odd*-harmonic distortion.

SINGLE-ENDED AMPLIFIER

In a single-ended amplifier, the harmonics that are generated are mainly *even* harmonics. On the other hand, a push-pull output stage usually operates in such a way that the transfer curve of one transistor in the output stage overlaps and cancels out the non-symmetrical nonlinearities in the other transistor of the push-pull stage. As a result, the even harmonics are largely canceled out. Most of the distortion then consists of odd harmonics alone.

Some harmonics are more displeasing to listeners than are other harmonics. In general the lower-order harmonics (say the second through the fifth) result in tones that are on the musical scale; hence they are not unpleasant to hear. On the other hand, the higher-order harmonics (say the seventh through the twenty-fifth) are mostly not on the musical scale and are decidedly unpleasant to listen to, even when the harmonics are fairly low in amplitude.

For example, assume we apply a 250 Hz sine wave to an audio system. If the system introduces harmonic distortion, we will find that the second through the sixth harmonics are musically related to the fundamental, hence they are not unpleasant to listen to although they certainly constitute distortion because they were not present in the input waveform. Additional musically related harmonics include the eighth, tenth and twelfth, as well as the sixteenth, twentieth and twenty-fourth. Non-musical, dissonant harmonics are the seventh, ninth, eleventh, thirteenth, fourteenth, fifteenth, seventeenth, eighteenth, nineteenth, twenty-first, twenty-second, twenty-third and twenty-fourth.

Although practically no music produced by acoustic rather than electronic musical instruments and practically no speech is purely sinusoidal, or lacking in harmonics, we do not want our electronic audio equipment to generate the harmonics. The job of the equipment is to duplicate the original input without introducing any harmonics of its own. The perfect amplifier, then, is one that reproduces exactly the waveform, no matter how complex, that is applied to it.

TOTAL HARMONIC DISTORTION

The harmonic-distortion factor of a signal is the ratio between the total rms values of all the harmonics to the total rms value of the fundamental plus all the harmonics. Expressing this factor as a percentage (multiply the factor by 100) gives us a measure of the percentage of *total harmonic distortion* (thd).

To be more exact, the percentage of thd is equal to the square root of the sum of the squares of all the harmonics divided by the square root of the sum of the squares of the

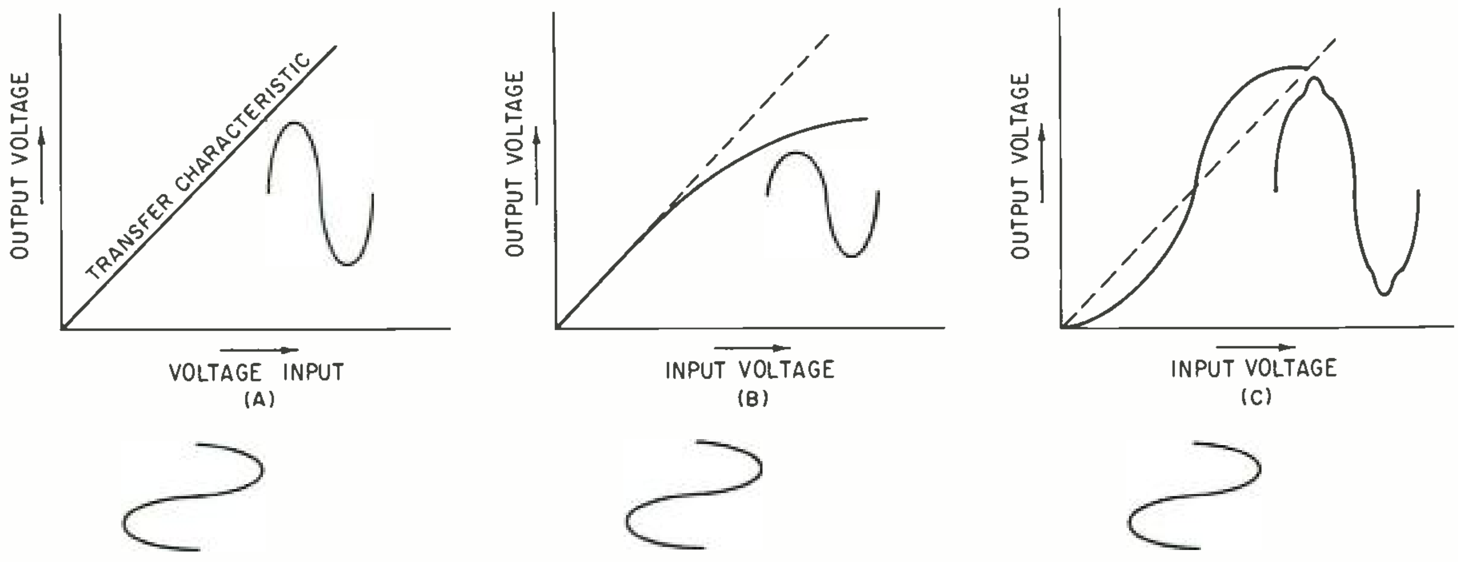


Figure 3. Input and output waveforms with various types of nonlinearities.

fundamental and the harmonics, all multiplied by 100.

Assume we have a distorted waveform (fundamental plus harmonics) with an rms value of 50 volts and we find that we have, in addition to the fundamental signal, a second harmonic of 2 volts and a third harmonic of 1.5 volts. Our percentage of thd is the square root of 2^2 plus 1.5^2 or 2.5 divided by 50, all multiplied by 100, or 5 per cent thd.

Sometimes the distortion is *weighted* in proportion to the order of the harmonics. When this is done, the percentage of the individual harmonics is multiplied by a weighting factor that increases as the order of the harmonic increases.

With just about every system of amplification other than class B, the percentage of total harmonic distortion decreases as the power output level is reduced. In addition, as the output power level is reduced the percentages of the higher order harmonics decrease more rapidly than those of the lower order harmonics. This means that the thd usually decreases as the power output is reduced. However, in some transistor amplifiers, when you go down to very low output powers, thd may actually begin to rise again slightly.

When negative feedback is used, all the harmonics are reduced in the same proportion. This does not affect their relative importance, except when the overload point is reached.

HOW MUCH DISTORTION?

An important question is just how much distortion we can perceive or tolerate in an audio system. We can, in general, tolerate less distortion with music than with speech. Also, as we increase the range of frequencies that our system covers, we can tolerate less distortion. This means that in a fairly restricted bandwidth system, say covering a frequency range of 100 to 5,000 Hz, we can tolerate far more distortion than in a wide-band audio system that covers from 20 to 20,000 Hz. The narrow-band system simply does not respond to the higher order harmonics. However, the disadvantage of the narrow-bandwidth system is that it does not respond to the desirable and necessary very low and very high frequencies. Hence, the price that we have to pay for increased frequency coverage is that we must exert more effort to acquire lower distortion in the system.

It is difficult to set down specific limits for the total harmonic distortion percentage. This is because the thd usually lumps together all the harmonics and does not specify which harmonics are involved.

As an example, suppose we consider a waveform with a thd of 5 per cent, as calculated above. Nothing is usually said about whether this percentage represents mainly odd harmonics, even harmonics or a combination of both odd and even. Further, if the harmonic distortion consists of a number of harmonics, as is usually the case, nothing is indicated in the percentage figure to tell us the relative amplitudes of the various harmonics making up the distortion. Any of these conditions would produce a differently shaped distorted waveform and a different effect on the listener.

In general, listeners will tolerate a much larger amount of even-harmonic distortion than odd-harmonic distortion. This is because the even harmonics are largely musical and nondissonant, while the odd harmonics are not musical and are dissonant. Some circuits and even some transistors are more prone to emphasize certain harmonics than others. As mentioned above, harmonic distortion in most push-pull stages is largely odd harmonic, the even harmonics being canceled out.

Because of these and other variables, the thd certainly does not tell the entire story of system performance. However, it does provide us with a simple, convenient, and easily duplicated test that allows us to compare one with another.

A number of tests were conducted some years ago by

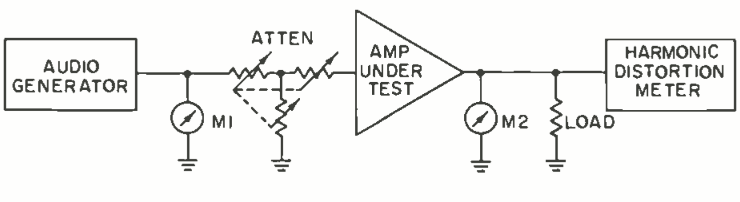


Figure 4. Test setup used to measure total harmonic distortion of amplifiers.

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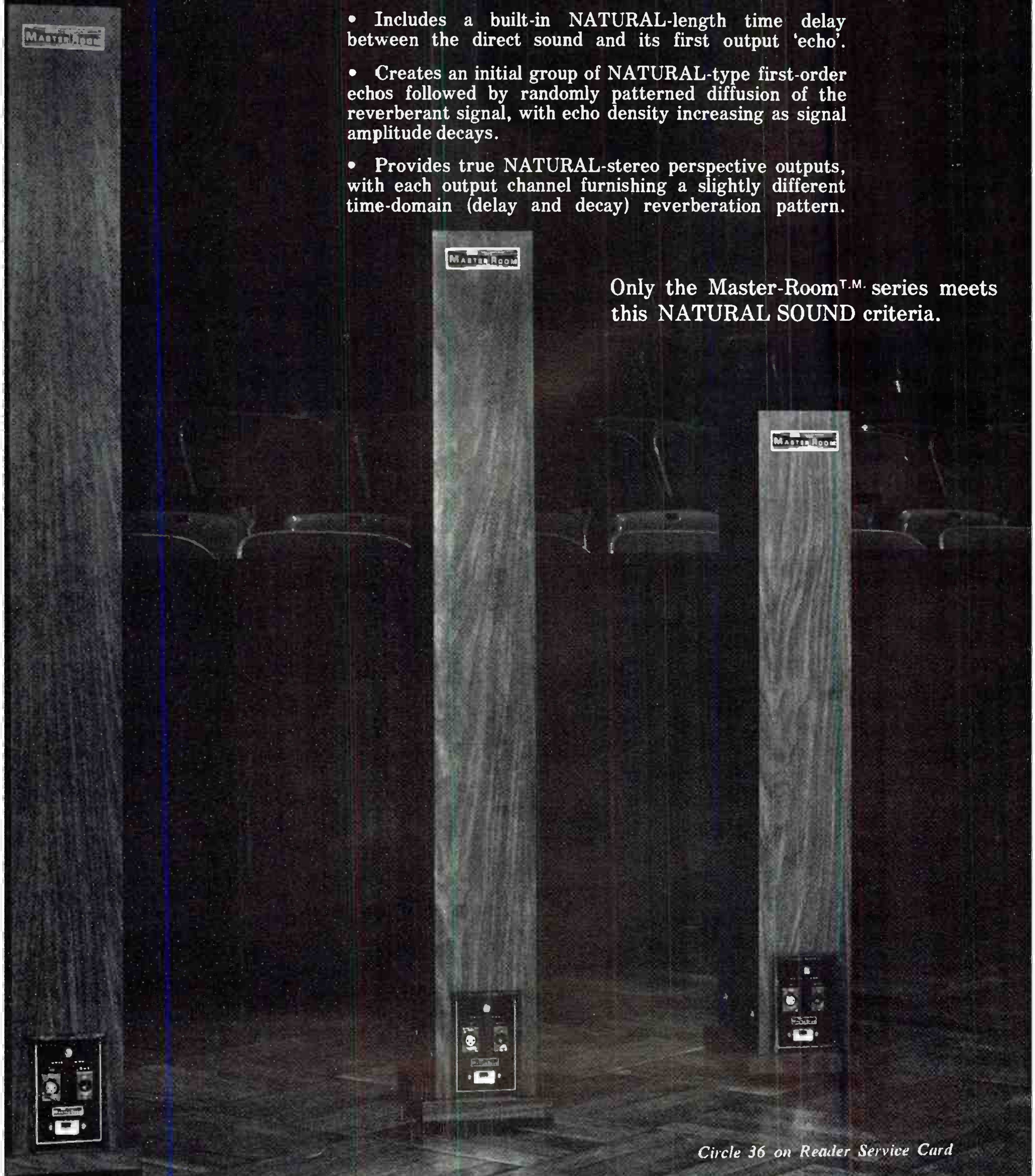

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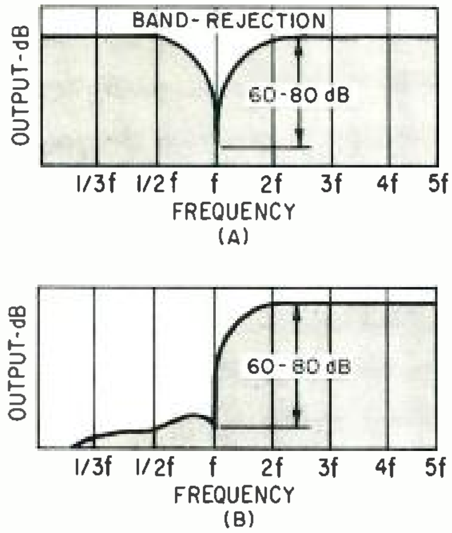


Figure 5. The two types of response curves used in harmonic distortion analyzers.

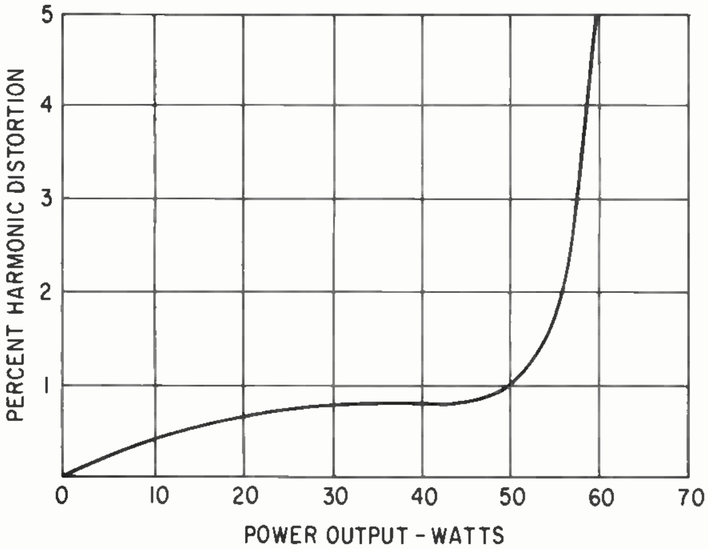


Figure 6. Distortion characteristic of a typical hi fi power amplifier.

Dr. Harry Olson of RCA using single-ended, low-power (3 watt) triode and pentode tube amplifiers. The tests were conducted in a typical living room environment with a noise level of about 25 dB. A limited number of critical observers were used to rate the intentionally introduced distortion from objectionable—through tolerable—to perceptible.

The results of these tests are the following: For an amplifier high-end cutoff of 7,500 Hz, objectionable distortion occurred at 4 to 4.8 per cent thd for music and 6.4 to 6.8 per cent thd for speech. Tolerable distortion occurred at 3.2 to 4.4 per cent for music and 4 to 4.8 per cent thd for speech. Perceptible distortion occurred at 0.95 per cent thd for music and 1.15 to 1.2 per cent thd for speech.

Next, the high-end cutoff was extended to 15,000 Hz. Under these conditions, objectionable distortion was 2.0 to 2.5 per cent for music and 3.0 to 4.4 per cent for speech. Tolerable distortion occurred at 1.35 to 1.8 per cent for music and 1.9 to 2.8 per cent for speech. Perceptible distortion occurred at 0.7 to 0.75 per cent with music and 0.9 per cent with speech.

Other tests, made with telephone line equipment by the British Post Office in conjunction with the BBC, disclosed the following results for *just detectable* second- and third-harmonic distortion: For second-harmonic distortion, up to 25 per cent below 100 Hz, up to 3 per cent below 200 Hz, up to 1 per cent below 400 Hz, and below 1 per cent above 400 Hz. For third-harmonic distortion, just detectable distortion was up to 5 per cent below 100 Hz, up to 2 per cent below 200 Hz, and up to 1 per cent above 400 Hz.

The better the equipment, the less thd it will have. A perfect amplifier would have zero per cent total harmonic distortion. Since such an amplifier does not exist, we will have to settle for a thd figure below 1 per cent for very good performance and below 0.5 per cent for exceptional performance. There are a few amplifiers available whose thd approaches or is lower than the residual distortion in the instruments being used to measure the thd.

MEASURING THD

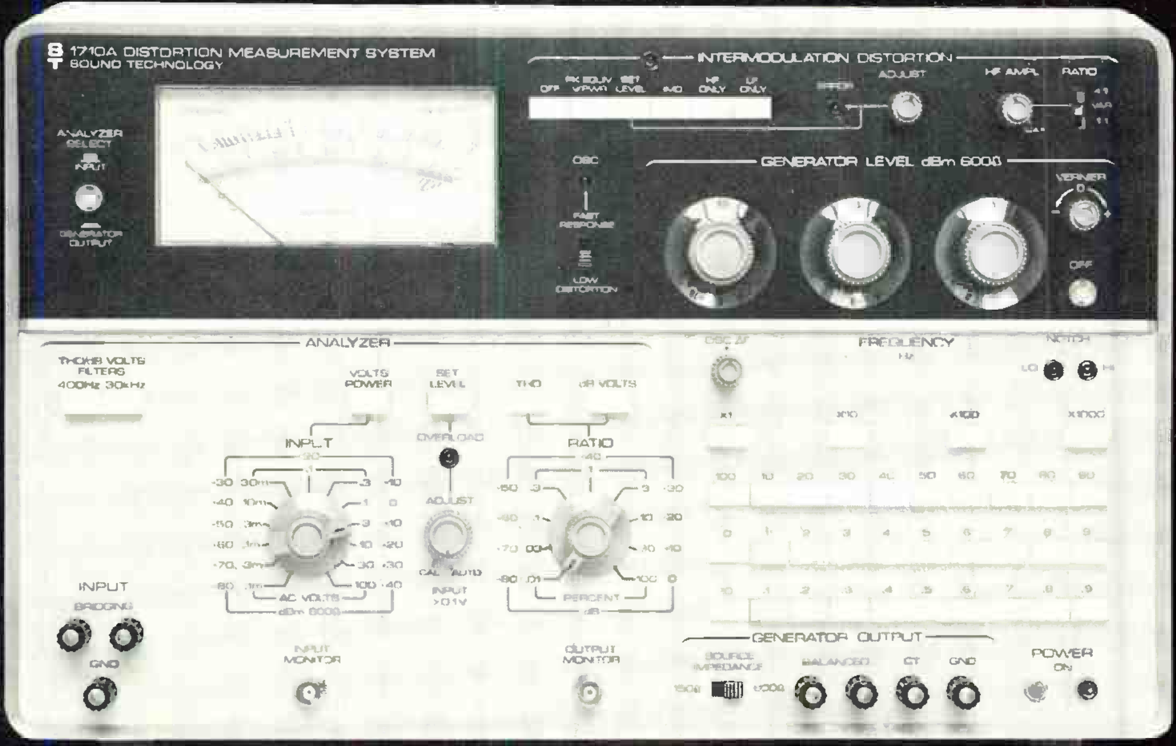
In order to measure total harmonic distortion we can use the test setup shown in FIGURE 4. A low-distortion audio generator, whose output is monitored with an external meter M1 (if the generator itself does not have such a meter), is applied through an attenuator to the input of the audio unit under test. In this case, we are showing an amplifier being tested. The output of the amplifier is monitored by a second meter, M2. The amplifier is also properly terminated by a load resistor as shown. The amplifier output is applied to the input of the harmonic-distortion meter. This meter, or analyzer, will read the percentage of thd directly

The harmonic-distortion meter contains a selective audio voltage amplifier, with adjustable attenuation, whose output is connected internally to a high impedance vacuum tube or solid-stage voltmeter circuit. The purpose of the selective amplifier is to suppress or null out the fundamental frequency of the audio generator so that a measurement can be taken of the remaining harmonics.

The usual method of obtaining this selectivity is by the use of a tunable Wien-bridge or bridged-T network that is used to put a sharp notch in the instrument's response curve (FIGURE 5A). When the notch is adjusted to the fundamental frequency of the audio generator, the fundamental frequency is effectively removed or suppressed by 60 to 80 dB. The meter in the harmonic-distortion analyzer now has applied to it all the other components of the waveform. These are mainly harmonics, but also included is any hum or noise produced by the unit under test.

When the instrument is used to take a measurement of thd, the selective amplifier is first bypassed entirely and a meter reading is taken of the output of the amplifier under test. This reading is the value of the fundamental frequency plus the distorting harmonics. The meter is now adjusted for full-scale (100 per cent) reading. Now the selective amplifier is switched in and it is carefully balanced to suppress the fundamental. This is done by adjusting the notch frequency of the selective amplifier for a null or minimum reading. Now the residual meter reading is a direct indication of the percentage of total harmonic distortion.

Some harmonic-distortion meters employ sharp cutoff high-pass filters to eliminate the fundamental frequency (FIGURE 5B). With such a curve, not only is the fundamental frequency removed but the hum and noise below the fundamental are also effectively eliminated. As such filters are not usually adjustable, these instruments may use a half dozen or so filters with different cutoff frequencies in order to permit measurements to be made at various fundamental frequencies. In some cases, two such filters are used, one cutting off at 400 Hz and the other at 1,000 Hz. This permits thd measurements to be made at



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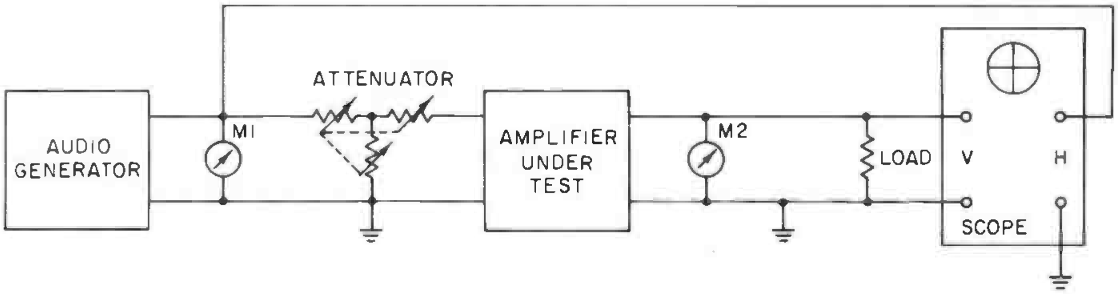


Figure 7. Test set up for oscilloscope observation of harmonic distortion.

these two fundamental frequencies.

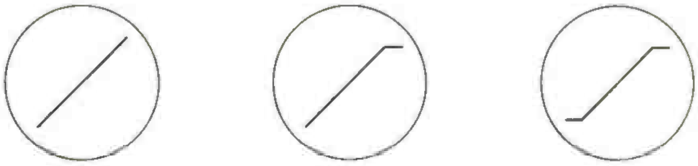
The measurements taken with these two types of harmonic-distortion meters will indicate two slightly different thd figures. The instrument with the band-rejection filter will usually yield slightly lower values of thd because its figures do not include the low-frequency noise and hum. In most cases it is the bridge-type unit that will be used.

A number of years ago it was common to take harmonic-distortion measurements at only one or two frequencies around the middle of the audio band, such as 400 or 1,000 Hz. Later, though, with improvements in audio gear and the emergence of really high-quality hi-fi consumer products, many manufacturers were anxious to show just how good their equipment was. Therefore, it became common to make thd measurements throughout the entire range from 30 to 15,000 Hz or even from 20 to 20,000 Hz. Such measurements impose a severe test on a unit because it is far more difficult to handle the very low and very high frequencies with a minimum of distortion than it is to handle the frequencies in the middle of the audio range.

It is also common to make thd measurements over a wide range of output powers or output voltages, from some very low value up to and beyond the overload region of the equipment under test. In general, as the output power or voltage is increased, so is the amount of distortion. Usually, the increase is smooth and gradual up to the overload point, where there is a sudden increase in distortion. Amplifiers should be rated at a power or voltage just below this overload point, while the thd is still only a small value, say 1 per cent.

In FIGURE 6 we see an amplifier's percentage of thd (at some mid-frequency) plotted against its output power. In this case, the amplifier has a thd of less than 1 per cent below output powers of 50 watts. At 50 watts, the thd is just 1 per cent. Above this power, the thd rises quickly to a value of 5 per cent at 60 watts. This amplifier would then be rated at 50 watts with 1 per cent thd.

A high-quality 50-watt amplifier whose thd has been measured over the entire audio range may have the following specification: "Total harmonic distortion at 1 per cent or below from 20 to 20,000 Hz within 1 dB of 50 watts." Such an amplifier would have no trouble at all in producing up to a full 50 watts of output at 1 per cent or less of thd over most of the audio range. At the very ex-



(A) (B) (C)

Figure 8. Distortion patterns on an oscilloscope being used to observe harmonic distortion.

tremes of the range, however, it would still be producing 1 per cent or less distortion at powers up to 40 watts (which is 1 dB below 50 watts).

Another unit might be rated as follows: "Total harmonic distortion below 2 per cent from 30 to 15,000." Such equipment might very well have a thd of a fraction of 1 per cent at 1,000 Hz, but distortion would not exceed the 2 per cent figure at full rated power output over the frequency range specified.

USING A 'SCOPE TO OBSERVE DISTORTION

An oscilloscope may also be used to observe harmonic distortion in an amplifier or other piece of audio equipment. As a rule, it is difficult to see distortion much less than 3 to 5 per cent on a 5-inch cathode-ray tube. In some cases, though, especially with higher order harmonics, some 2 to 3 per cent distortion can be observed. It is important that the 'scope used have vertical and horizontal amplifiers with similar or equal frequency responses and phase characteristics.

To check distortion with a 'scope, use he setup shown in FIGURE 7. Here the audio generator is fed through a monitoring meter and an attenuator to the input of the amplifier under test. The output of the generator is also applied to the horizontal-input terminals of the 'scope, whose horizontal sweep frequency is turned off. The output of the amplifier under test is monitored by meter M2 and is properly terminated in a load resistor. This output is also applied to the vertical-input terminals of the oscilloscope. The frequency of the audio generator is usually set at some convenient middle frequency, such as 400 or 1,000 Hz, or it may be set to some low frequency, such as 40 or 50 Hz.

The output of the generator is now increased from some very low value up to the point where the amplifier begins to overload or up to the rated power output.

The waveform seen on the 'scope will be a perfectly straight diagonal line, assuming that the 'scope's gain controls are adjusted so that equal voltages are applied to the deflecting plates of the cathode-ray tube (FIGURE 8A). This straight line is actually the transfer characteristic of the amplifier under test. At and above the overload point the straight line will begin to show some curvature at either one or both ends, or it may show some curvature somewhere along the length of the trace. If the curvature is at one end (FIGURE 8B), you are seeing the results of even-harmonic distortion, and if the curvature is at both ends (FIGURE 8C), you are seeing the results of odd-harmonic distortion.

A drawback of this technique is that you cannot readily determine the actual percentage of harmonic distortion. About all that can be done is to determine whether or not distortion is present. The more the nonlinearity or distortion, the greater will be the curvature of the scope trace.

A measurement of total harmonic distortion, although it does not tell the entire story about a unit's characteristics, is one of the most useful performance specifications that can be measured. ■

HARALD BODE

Frequency Shifters For Professionals

Integrated into electronic synthesizers, frequency shifters increase the possibilities of tone interplay and innovation.

IN CONTRAST to transposing devices, frequency shifters change the harmonic structure of any natural or synthesized sound received at the input, thus creating new sounds for the innovative user. Among the different types of frequency shifters known, the model 735 Bode Frequency Shifter and its counterpart, made by Moog Music, Inc. are the most versatile. In these devices, the amount of frequency shift is voltage-controlled according to one of two control modes.

In the linear mode, the amount of shift is continuously variable from +5 kHz, through zero, to −5 kHz. In this mode of operation, an alternating control voltage introduces additional sidebands into the shifted outputs without actually changing the average amount of shift. In the exponential mode, the amount of frequency shift doubles for each one-volt increase in control voltage, thereby producing changes in the amount of frequency shift that run parallel to the frequencies of synthesizer oscillators and filters that are being controlled from the same voltage. The resulting effects and some interesting applications will be described in this article.

Frequency shifters have been built for a number of purposes, such as the reduction of acoustical feedback (howl) in sound reinforcement systems and, in a multiple single sideband configuration, for the simulation of a choral tone effect. Whereas instruments of this kind use small frequency changes to achieve the desired results, there is an apparatus used for substantial changes of musical frequencies. This is known in the German broadcasting system under the name *Klangumwandler,* which, directly translated, means *sound converter*. This device operates through double heterodyning and single sideband production through the use of a single sideband filter.

Harald Bode is a well-known inventor of devices used in electronic music. He heads Bode Sound Company in North Tonawanda, N.Y.

The techniques employed for frequency shifting are basically the same as known for single sideband production—heterodyning and the use of the phase shifting principle. I used a combination of both principles in a special frequency shifter built for the Electronic Music Centers of Columbia and Princeton universities in 1963.[6]

Since the introduction of this rather specialized instrument, I have developed a number of different models. Among these is a carrier injection model of 1964, which subsequently was manufactured by Moog. In a more recent joint effort by Moog and myself, a versatile frequency shifter was developed and presented at the AES Spring convention in Los Angeles in 1972.[7] This model has, among other features, a built-in beat frequency quadrature oscillator (patented in 1974), which is voltage controllable, including a linear-to-exponential interface, making this frequency shifter compatible with voltage controlled synthesizer modules and functions.

BEAT FREQUENCY QUADRATURE OSCILLATOR

A basic (simplified) block schematic diagram of this frequency shifter is shown in FIGURE 1. Through the input terminal, the program signal is entered into two phase shifting networks, ϕ_1 and ϕ_2, which produce two output signals with a 90 degree phase difference relative to each other[8]. These phase-shifted signals are then fed to the first inputs of two four quadrant multipliers. The second inputs of the same multipliers reecive two 90 degree out-of-phase signals from a beat frequency oscillator, which is composed of a fixed (20 kHz) and a variable (15-25 kHz) oscillator. The fixed oscillator is followed by a gate, a low-pass filter (or resonance circuit) to secure pure sine waves, and by two phase shifters ϕ_3 and ϕ_4, which produce two 90 degree out-of-phase outputs (sine/cosine relationship). After mixing these two components of the fixed frequency with the variable frequency, two beat frequencies are obtained at the outputs of mixers 1 and 2, which again are in sine/cosine relationship. At the mixer outputs, the 10 kHz low pass fil-

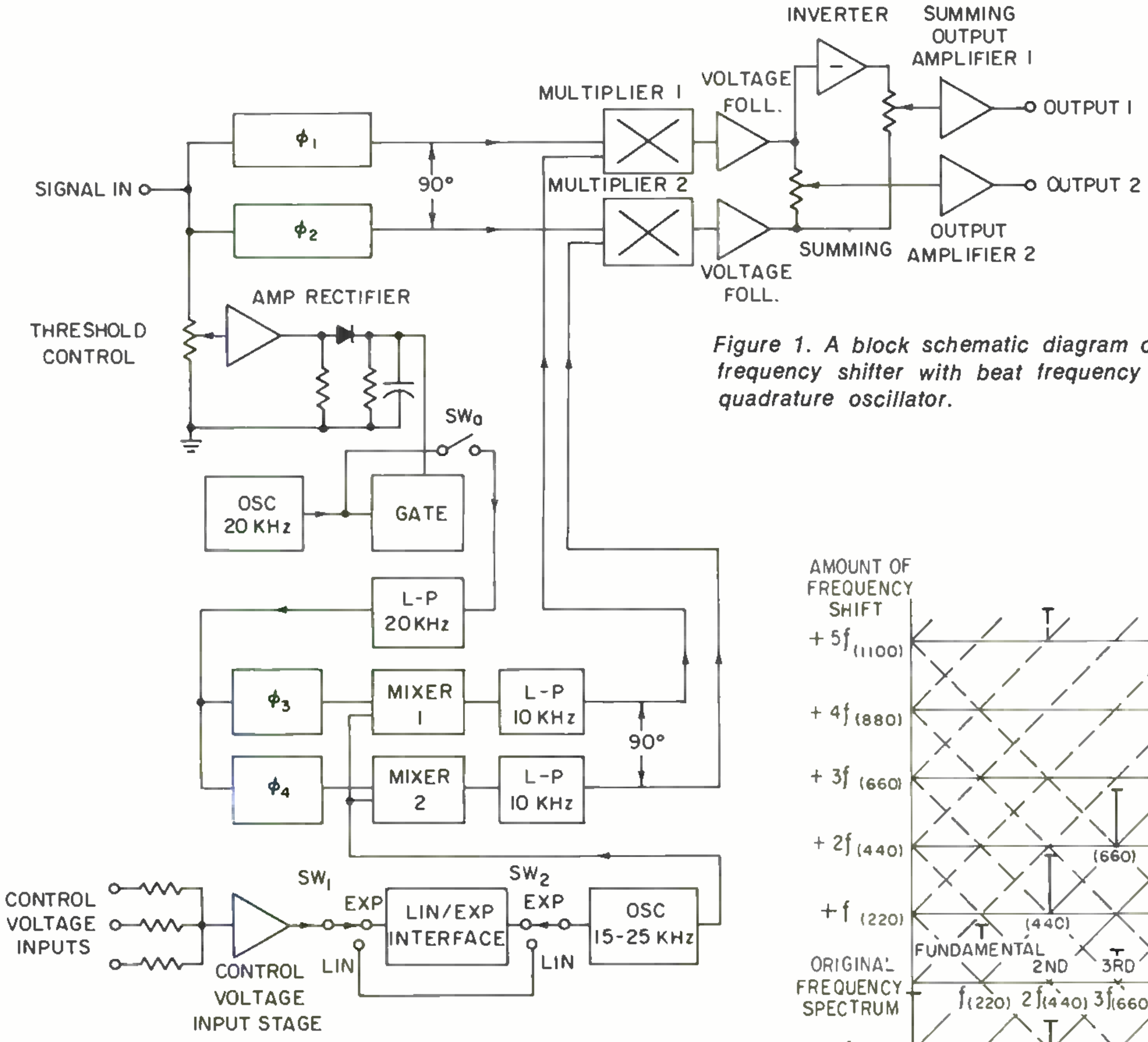


Figure 1. A block schematic diagram of frequency shifter with beat frequency quadrature oscillator.

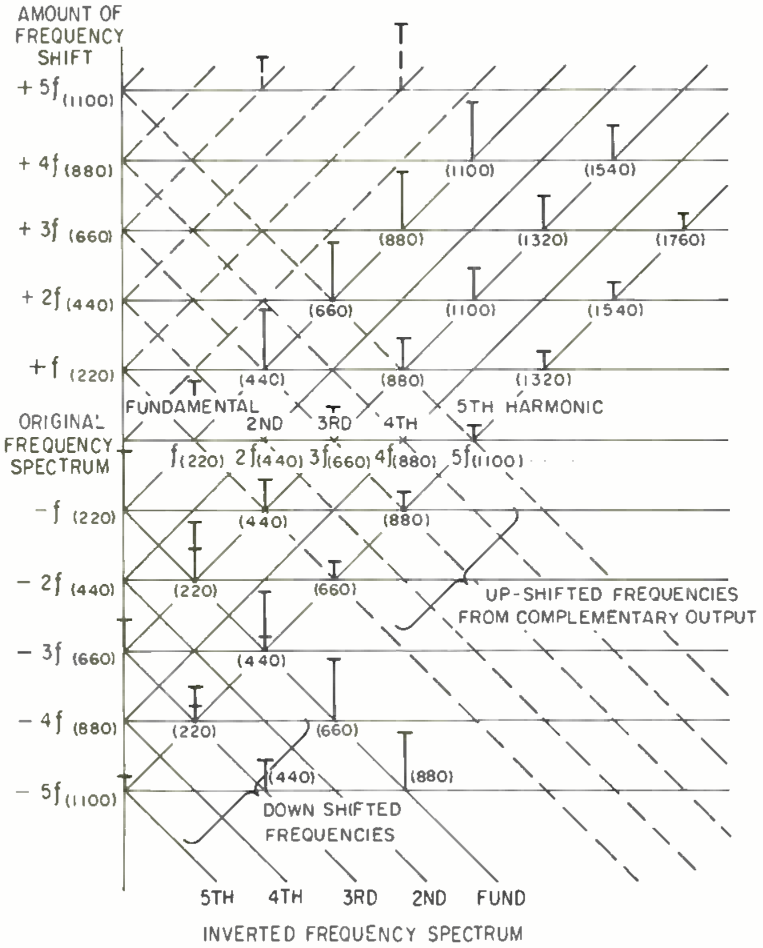


Figure 2. The change of harmonics through frequency shift.

ters are provided to eliminate the high frequency components of the beat frequency oscillators. Through the use of direct-coupled circuitry, this oscillator operates with a constant amplitude from d.c. to the highest beat frequency.

When displaying the two output components on the X and Y axis of an oscilloscope, a clean circle appears, which reverses its rotation when going through zero beats.

Going back now to the four quadrant multipliers 1 and 2: These produce two sidebands each with a suppressed carrier. The sidebands are made up of the beat frequency plus and minus the program frequencies received at the input. Due to the phase relationship between the two multiplier outputs, one of the sidebands is cancelled when combining the two output signals (at the voltage follower outputs) through summing. When combining the inverted output signal of multiplier 1 with that of multiplier 2, the other sideband is cancelled. Thus the two opposite sidebands appear at outputs 1 and 2, which means that one output produces an up-detuned signal and the other a down-detuned signal.

FREQUENCY-SHIFTED SIGNAL

So far, I have discussed the basic performance of frequency shifting functions. Before going into a description of some of the other features of this instrument it may be of interest to see what the analysis of a frequency-shifted signal looks like.

As an example, FIGURE 2 gives a graphical display of the first five harmonics of a sound before shifting (original frequency spectrum) on the horizontal center line, and after shifting above and below this center line. The approximate amplitude values are chosen for a square wave, a waveform which has a hollow, clarinet-like quality because it has only odd harmonics.

Let's assume for the sake of illustration that the fundamental frequency f equals 220 Hz. Then the harmonic frequencies will be 2f = 440 Hz, 3f = 660 Hz, 4f = 880 Hz, and 5f = 1,100 Hz. If we now shift the frequency up by +f, or +220 Hz (seen on the vertical scale), then all of the harmonics go up in frequency and the new, shifted frequencies can be found at the intersection of the solid diagonal lines and the horizontal line identified by $+f$. In this case, the fundamental of the original sound has changed to 440 Hz, the third har-

Fig. 3. The front panel layout of the Model 735 Bode frequency shifter.

monic to 880 Hz, and the fifth harmonic to 1,320 Hz.

It can be recognized immediately that these new frequencies are the first three harmonics of a sawtooth wave (or stringlike quality), or, if not extended, of a flute tone, with a fundamental one octave higher than that of the original.

This is of course a very special example, which happens to represent a frequency change by an amount that equals the fundamental frequency of the original sound. If, in contrast, the frequency shift is not related to the frequencies of the original spectrum, a new sound is produced, the partials of which are no longer harmonically related. For instance, if the tone spectrum shown in FIGURE 2 is shifted by +50 Hz, then the fundamental changes to 270 Hz, the second harmonic to 490 Hz, the third to 710 Hz, the fourth to 930 Hz, and the fifth harmonic to 1,150 Hz, and the original sound loses its identity. Sounds of bells, chimes, carillons, and the like fall into the category of tones with non-harmonic structures. The frequency shifter is capable of producing an endless variety of sounds of this type.

From the discussion of up-shifted sounds, the reader may derive as well the structure of down-shifted sounds. One interesting feature of the down-shifting is that, depending upon the amount of shift, part of the original spectrum or all of it is inverted, which leads to another family of interesting new sounds.

In addition to the group of partials obtained from one of the outputs of the frequency shifter and represented by the solid lines in FIGURE 2, there are the partials derived from the complementary output of the instrument, represented by the dashed diagonal lines.

EXPONENTIAL SHIFT CONTROL

So far only a few examples for frequency shifting one single note have been discussed. But what if we have found an interesting sound and want to repeat it over

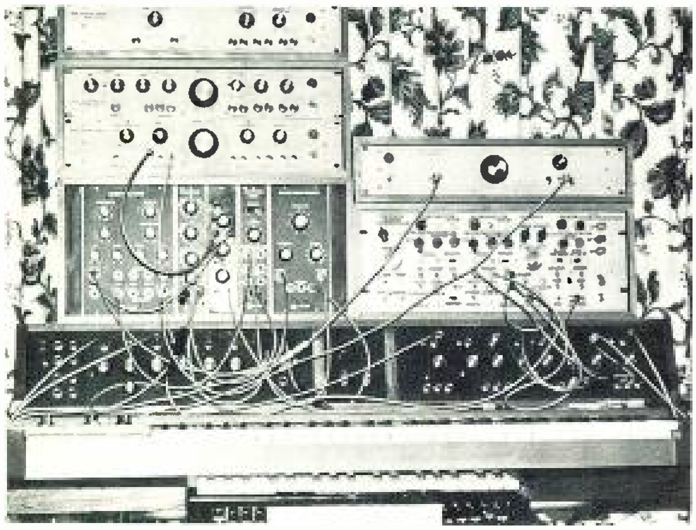


Figure 5. The author's synthesizer, using frequency shifters.

Figure 4. The Moog version of Bode frequency shifter (Model 1630).

the entire keyboard? This is accomplished with an exponential shift control.

Evidently the amount of frequency shift will need to be changed with the fundamental pitch so that the ratio of the shifting frequency *versus* the fundamental remains the same over the keyboard range. This will become obvious when considering the first example, in which the amount of shift equalled the fundamental frequency. From this it follows that the amount of shift or the b.f.o. (beat frequency oscillator) frequency of the shifter has to move in the same musical intervals as the audio frequency fed to the signal input. Since the frequencies of the keyboard scale follow an exponential function, the same has to be true for the oscillator frequency of the shifter.

For this reason, the linear voltage intervals of the keyboard controller (or ribbon controller) of a synthesizer have to be translated into exponential intervals for the local oscillator of the frequency shifter, as shown in the diagram of FIGURE 1 (linear to exponential interface), just as it is done on voltage-controlled oscillators and other keyboard-controlled synthesizer modules. Through the inclusion of the exponential mode, the frequency shifter becomes a real-time performance instrument within a synthesizer installation.

Another important feature is the variable sensitivity carrier squelch circuit, which eliminates the almost inaudible carrier feedthrough when the audio signal level at the input is below a preset threshold level.

FIGURE 3 shows the front panel layout. The threshold control for the squelch circuit is seen on the left hand side. A light emitting diode above the control knob lights up when the incoming signal is above the preset threshold level. A mode selector and scale switch, under the heading, *Scale*, facilitates the selection of the exponential mode and the ranges from +5 to −5 Hz detuning through +5 kHz to −5 kHz detuning (linear), as well as calibration mode, which operates in conjunction with the *zero adjust* control. With this control, the instrument is initially calibrated to zero beat, indicated on the l.e.d. above the control knob.

Using the main tuning knob (*amount of shift* control in the center of the instrument), the built-in beat frequency oscillator is either detuned in linear increments in accordance with the range selected on the *scale* switch or in exponential increments. In the latter case, the change by one dial increment corresponds to a one-octave frequency change.

The *mixture* control facilitates the mixing of the two up and down detuned signals in any desired proportion.

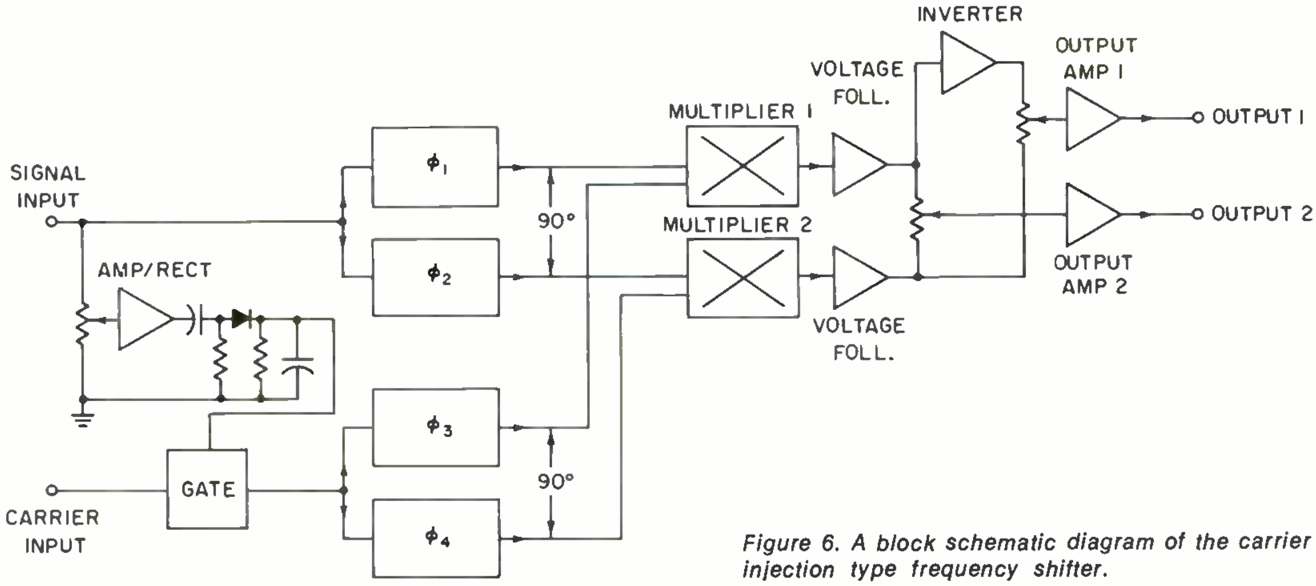


Figure 6. A block schematic diagram of the carrier injection type frequency shifter.

In the center position, the output A + B equals the performance of a ring modulator.

The inputs of the frequency shifter include one input jack for the program signal, *signal in*, and three input jacks for control voltages, *control inputs*. The outputs feed into two jacks each for one of the sidebands, *Out A*, two jacks each for the other sideband, *Out B*, and two jacks each for the mixture of both sidebands.

On the right hand side, the line switch and the pilot light is shown on this particular model, which is equipped with a built-in power supply.

FIGURE 4 shows the Moog version of the Bode frequency shifter, which fits into the modular assembly of the Moog synthesizers. All of the controls just explained (with the exception of the power switch and pilot light) can be found on the 1630 Moog model in a different geometric arrangement. Electrically both models are identical.

A limited size custom synthesizer is shown in FIGURE 5. Here the model 735 frequency shifter is in a case on the left hand side directly above the case with the Moog modules.

OTHER TYPES OF FREQUENCY SHIFTERS

Other types of frequency shifters include the heterodyning model, the carrier injection model, and models with a built-in quadrature oscillator. Of these, the latter two will be described briefly.

A block schematic diagram for the carrier injection model (Bode model 750) is shown in FIGURE 6. Here the incoming signal is fed to two phase-shifting networks, ϕ_1 and ϕ_2, the output signals of which are 90 degrees out of phase relative to each other over the audio range (35 Hz to 16 kHz). The outputs of these networks are connected to the first inputs of multipliers 1 and 2, the second inputs of which receive their signals from two phase shifting networks, ϕ_3 and ϕ_4, the basic circuit of which is identical to that of ϕ_1 and ϕ_2 with the exception that they cover a frequency range from 8 Hz to 4 kHz. This latter range is more meaningful for frequency shifting carrier frequencies.

The phase filters ϕ_3 and ϕ_4 receive the carrier (usually a sine wave) through a gate, which is opened at a preset level of the program signal, so that there is no carrier feedthrough in the quiescent state.

The output signals of multipliers 1 and 2 are summed at the voltage follower outputs to produce a frequency-shifted signal at output 2 in much the same way as it was described for the system in FIGURE 1. A signal of opposite shift direction is produced at output 1 by summing the inverted signal of multiplier 1 with the non-inverted signal of multiplier 2.

FIGURE 7 shows the front panel layout of the Bode model 750 carrier injection frequency shifter. The controls are, from left to right, the squelch threshold control with the l.e.d. above the control knob, the sideband switch, which facilitates sideband reversal, and the mixture control for mixing of the up-detuned and the down-detuned signal in any desired proportion. If the proportions are equal, the signal at the A + B output equals the performance of a ring modulator.

The inputs of this frequency shifter include one input jack each for the program signal (audio in) and for the carrier signal (sine wave, + 2 dBm nominal level). The outputs feed into two jacks each for the upper sideband (*A Out*, up-shifted signal), two jacks for the lower sideband (*B Out*, down-shifted signal), and two jacks for a mixture of both (*A + B*).

In the installation of FIGURE 5, this frequency shifter can be recognized on top of the instruments on the left hand side.

The block schematic diagram of a further frequency shifter type with a quadrature oscillator for producing the frequency shifting sine/cosine signals is shown in FIGURE 8. From the preceding descriptions, this schematic should be self-explanatory. The Bode model 741 frequency shifter uses this system for feedback suppression in sound reinforcement systems. In this application, the

Figure 7. The front panel layout of the carrier injection-type Bode frequency shifter.

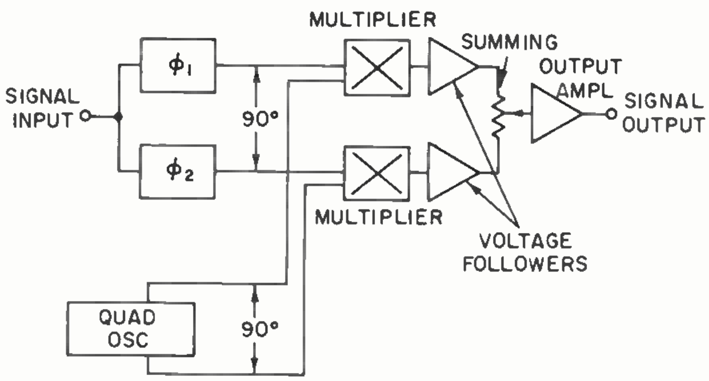


Figure 8. A block schematic diagram of frequency shifter with built-in quadrature oscillator.

quadrature oscillator provides a frequency shifting carrier in the range from 0.5 to 5.0 Hz. The front panel layout of this feedback suppressor is shown in FIGURE 9.

This instrument has also other interesting studio applications. For instance it can be used as a pseudo stereo and ambience effect enhancement device, supplying a complementary signal for a second channel when fed with monophonic program material at its input. In FIGURE 5, the frequency shifter is shown on top of the right hand equipment assembly.

A FEW TYPICAL APPLICATIONS

The typical applications of the model 735 can be put into four basic categories:

1. The simple up- and down-detuning of sounds, including passes through zero shift and production of "mirror image."
2. Frequency shift modulation around zero (or any other center frequency).
3. In-step detuning with voltage controlled synthesizer modules.
4. Repetitive detuning in tape loop. (Iteration effect).

Here are some typical effects which can be obtained. Triggering an envelope follower in conjunction with an envelope generator from a drum sound source (which also connects to the signal input) and feeding the voltage contour obtained from the envelope generator to the control input of the shifter will result in a varying frequency shift contour at the individual drum tone bursts (at a speed depending upon the decay time set at the envelope generator) and will yield a whole new class of sounds.

By setting the main tuning control to zero and applying a subsonic square wave to the control voltage input (linear mode), the up- and down-detuned outputs will switch places, resulting in a new type of special effect when heard over two channels. When this square-wave frequency is raised and enters the audio range, a completely new effect is obtained. In addition, a number of other effects will be produced with different types of wave shapes applied to the control input. A sine wave in the order of 5-6 Hz will result in a stereo vibrato. A sawtooth wave around 1 to 2 Hz will produce a somewhat dramatic effect. With pink noise applied to the control input, the program material will assume a hoarse quality which can be remixed with the original program signal.

By selecting the exponential mode and feeding the control voltage of the keyboard controller of a synthesizer into the control input of the frequency shifter, an infinite variety of new harmonic and non-harmonic sounds can be obtained when feeding the synthesizer tone signal into the signal input of the shifter. In this mode, the shifter becomes an integral part of the synthesizer, capable of being programmed into a large number of systems configurations.

A further special category of sounds obtained with the frequency shifter is the *iteration* effect, also referred to as the spiraling echo effect, which is produced by inserting the shifter in the line between the output of a recorder to its input. In this setup, the delayed sound received at the playback head is frequency shifted, then rerecorded, played back and frequency shifted again and again. An increasingly detuned sound is created, the character of which is determined by the amount of tape delay and the amount and sign of detuning. Evidently other delay devices can be used, such as digital delay lines, acoustical delay lines and the like.

The effects achieved with simple detuning of quasi-pitched sounds, such as drums, bells, and chimes cannot be overlooked; a frequency shifter can be a rather useful instrument with a drum section. Further applications include the processing of the human voice and many other natural as well as synthesized sounds.

The carrier injection model 750 can also be made into a rather versatile instrument by using a voltage-controlled oscillator (such as the Moog 921) for the carrier input. Almost all of the complex functions just described can be performed, with the exception of the frequency shifts through zero and modulation around zero shift. In the *exponential mode*, obtained through the Moog 921, which is controlled by the keyboard controller and which supplies the carrier frequency, very rich sounds can be obtained when using outputs other than sine waves, such as the triangle, square wave or sawtooth waves.

From these limited examples and from the preceding description it will become quite clear that a frequency shifter can be a most powerful tool for the production of new sounds. ■

Figure 9. A front panel layout of anti-feedback frequency shifter.

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NORMAN H. CROWHURST

Feedback, part 3

Interaction over two or more stages and special cases where amplification doesn't follow the rules are covered in the concluding study of feedback.

THE BASIC THEORY developed in the previous two parts is not too easy to apply directly. There is still a lot to cover. We could expand this to fill any number of parts; but instead we will try to give you the core of it all, in this third part. First, we'll recap the interaction concept that I developed back in 1953, then we'll take a look at some of the things that theory overlooks.

INTERACTION CONCEPT

A single stage of amplification will always have one high frequency roll-off and, if it uses capacitive or inductive coupling, it may also have a low frequency roll-off. FIGURE 1 shows a roll-off that occurs, in the absence of feedback, at 1000 Hz. If it occurs at 10 kHz or 100 kHz, or anywhere else, the effect is similar, just at a different place.

Now, 6 dB of feedback pushes the 3 dB point up an octave. 12 dB of feedback will push it up two octaves, and so on. If you look at a low frequency roll-off the effect is similar: each 6 dB of feedback will push the turnover point down one octave. If it is direct-coupled, it goes down to zero frequency anyway.

That is, in a sense, the starting point for the interaction concept. When you move to using feedback over two or more stages, the extension of frequency response is no longer so simple, but involves interaction between the stages, brought about due to phase effects. FIGURE 2 shows the effect of two stages, each having a roll-off shown by curve (A).

The two combined produce the response at curve (B), before any feedback. First, note that, as you add feedback, while you extend the frequency response where it is level, just as you did with one stage (FIGURE 1), the ultimate roll-off does not change. Because you cut gain, you reach that roll-off later. This ultimate roll-off is represented by line (D).

Now the 6 dB/octave. or half slope point, slides down a 6 dB/octave line, (C), as feedback is changed. Here, with two identical roll-offs at 2 kHz, each 6 dB of feedback extends the point of contact with the 6 dB/octave line by half an octave, which means it drops 3 dB in absolute level or, referred to the gain level with feedback, it rises 3 dB.

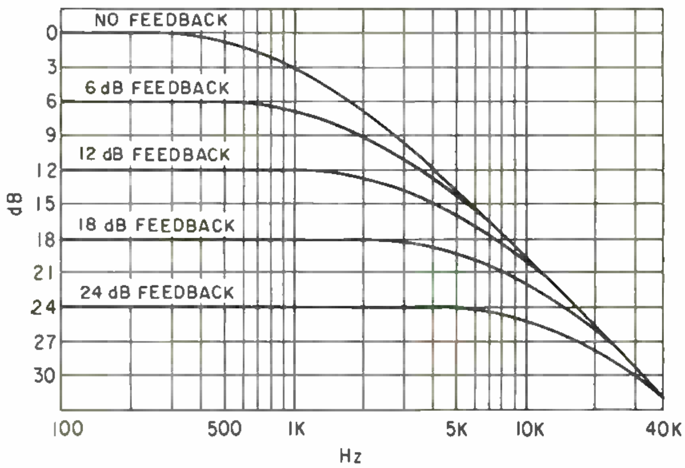

Figure 1. Effect of feedback when only one roll-off is effective. (From Crowhurst, N. H. **Feedback***, 1952).*

Without feedback, this point is 6 dB down, at 2 kHz. With 6 dB feedback, it is 9 dB down at 2.828 kHz or, relative to the new level with feedback (curve E) 3 dB down at that point. This point is of interest in active filter design. Another 6 dB of feedback makes the touch point 12 dB down at 4 kHz or, relative to the level with feedback, it is zero dB at 4 kHz, with a peak of about 1.25 dB at 2.828 kHz.

Going to 18 dB of feedback pushes the touch point up another half octave, but now it is up 3 dB from the level with feedback, with a peak of about 3.6 dB a little below that frequency. The last one shown, with 24 dB feedback,

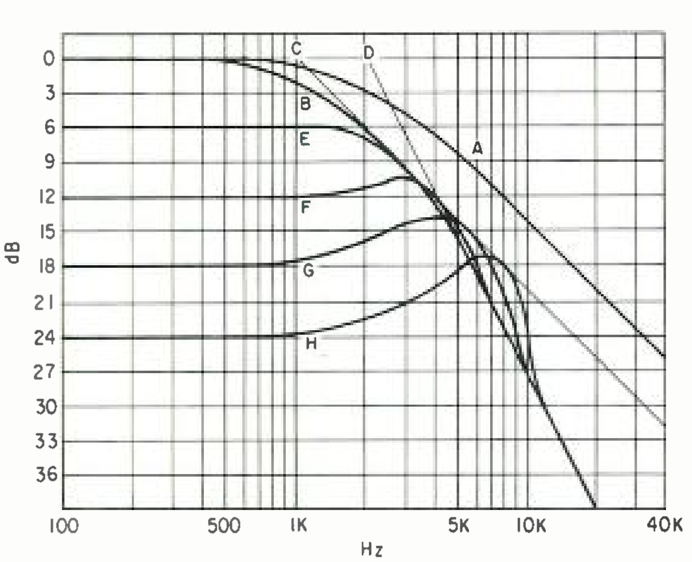

Figure 2. Effect of feedback with two identical roll-offs at the high end. (From Crowhurst, N. H. Feedback, 1952).

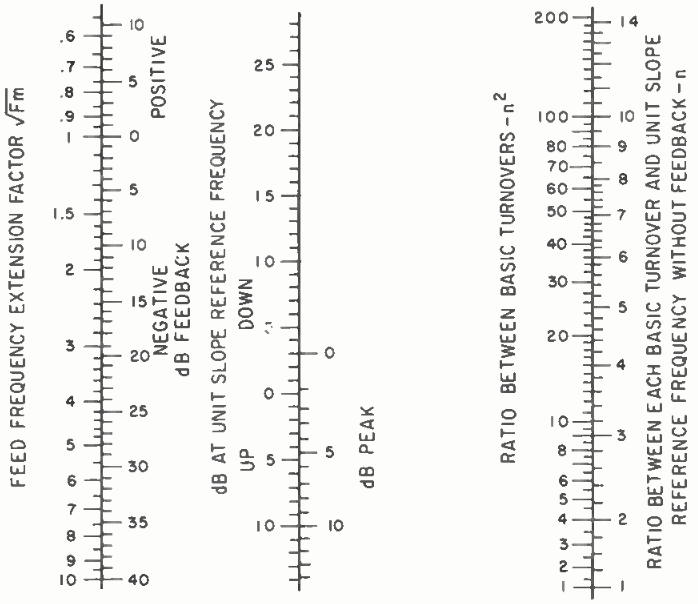

Figure 3. An abac for calculating response details in feedback over two roll-offs. (From Crowhurst, N.H., High Fidelity Sound Engineering, 1961).

makes the touch point up 6 dB, from level with feedback, at four times the original frequency, or 8 kHz, and about 6.3 dB peak just below that.

At the low frequency end, this pattern is exactly reversed when you have two elements contributing to a low frequency roll-off within a feedback loop. FIGURE 18 is an abac to facilitate calculating what feedback does in all such cases. In FIGURE 2, we assumed identical roll-offs. That condition is a special one, which will not often apply. FIGURE 4 shows how to apply FIGURE 3 in locating the response.

You have two turnovers, spaced at a frequency ratio of n^2. Midway between them, the response without feedback will have a slope of 6 dB/octave. Applying feedback will extend the touch point on the 6 dB/octave (or unit slope) by the extension factor, read across the left hand line in FIGURE 3. The chart can be used to find all reference points on the curve, given the necessary data.

MORE THAN TWO STAGES

Where feedback covers more than two stages, which is becoming more rare in these days of solid state circuitry, complete analysis becomes more complex, because the roll-offs can come in all kinds of combinations, not just a simple ratio. But we can simplify this by taking a best case, which under other circumstances can become a worst case.

If you have three roll-offs within a limited range of roll-off points, say a 10:1 frequency range, the best case, for achieving the most feedback without peaking first, and becoming unstable second, is when one roll-off acts first, say at 10 kHz with the other two acting at the other extreme, in this case 100 kHz. With that combination (FIGURE 5), 11 dB will reach the peaking boundary (*i.e.* more than 11 dB will cause a peak in the response) and 28 dB will make the circuit go into oscillation.

From another viewpoint, that is a worst case. If the first two rolloffs to act are on a 10:1 ratio, the worst case is when the third one is also at the second of these frequencies. Putting the third one beyond the second frequency will improve the figures slightly, but not much, unless the third frequency is removed very much further out.

FIGURE 5 tells the limits, but it does not tell what happens in between, for which FIGURE 6 is helpful. To use this, you take the peaking boundary given by FIGURE 5 as the amount of feedback allowable without peaking. *dB Excess Feedback* on FIGURE 6 is the amount of feedback more than this. Thus if n is 10, as in the example of the previous paragraph, excess feedback starts at 11 dB, and 20 dB feedback would be 9 dB excess feedback, resulting in between 5.2 and 7 dB peak, probably about 6 dB.

If the first roll-off is at 10 kHz with identical rolloffs, the peak would be at a little over 12 kHz, and with very large roll-offs, it would be at about 7 kHz. With the 10:1 ratio it would be somewhere between these extremes.

Earlier presentations extended these predictions to as many as five roll-offs within a feedback loop. This shows the method. If anyone wants me to go further with this, we can pursue it later. In the meantime, let us look at other aspects of feedback.

WHAT THE THEORY DOESN'T SHOW

All that theory is based on analyzing feedback performance, using frequency as a reference. It assumes that all components behave essentially the same throughout the waveform cycle. That is, amplification is essentially linear, and impedances due to dynamic active components do not change during the audio cycle.

For class A amplification, that may be a reasonable assumption. But what about when an amplifying device runs into clipping or cut-off? Then, although it is not behaving quite like a digital device, it does have two quite definite states during the audio cycle. For part of the cycle it is amplifying and for part of it, it isn't.

What about class B amplification, where two transistors share the complete waveform, so that, when one isn't amplifying, the other one is? That may be okay for many

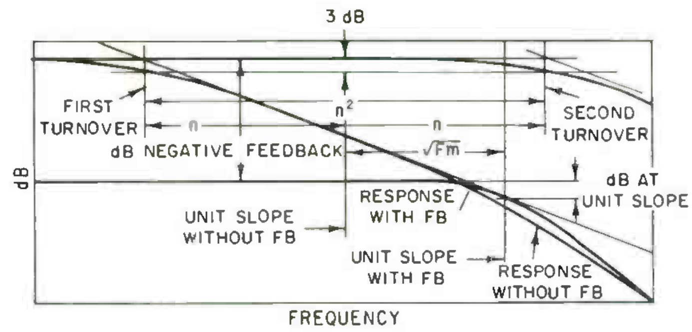

Figure 4. How the abac of Figure 3 is used. (From Crowhurst, N. H., High Fidelity Sound Engineering, *1961).*

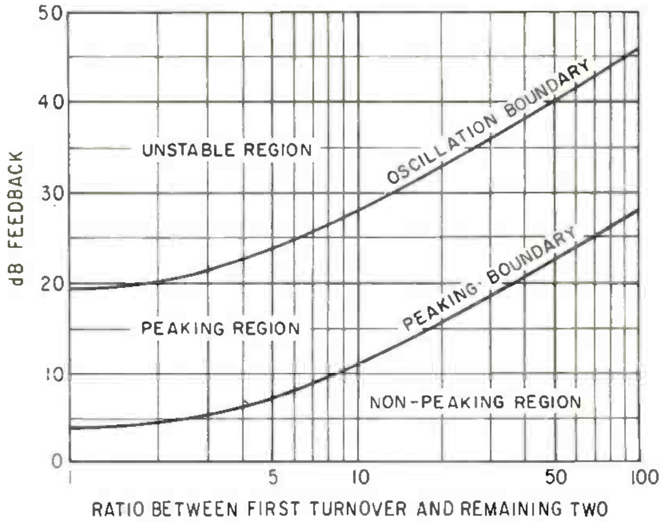

Figure 5. Chart giving boundary conditions for feedback over three stages of roll-offs. (From Crowhurst, N. H., High Fidelity Sound Engineering, *1961).*

purposes, provided the transistors match up, that is, provided that each starts to amplify at precisely the same point on the waveform as the other leaves off. Otherwise you have a two-state system.

Mathematical or theoretical analysis doesn't work too well anymore. Nor does the digital form of two-state. Now you have to treat the system qualitatively on a time-based analysis, considering what it does while the devices operate at their normal amplifying level, and what it does during those parts of the audio cycle where the amplification quits.

One example of this can occur in the relatively simple single stage transistor amplifier, discrete. Look at FIGURE 22. While it is amplifying, the bias current of the second stage holds it conducting, and the output side of the coupling capacitor is, say, a few hundred ohms from ground. Its input side couples from the collector of the preceding stage, whose maximum impedance is the value of the collector resistor, say 10 kΩ. The following stage bias resistor is, say 560 kΩ.

But because, while the transistor is conducting, that side of the capacitor sees a few hundred ohms to ground, the

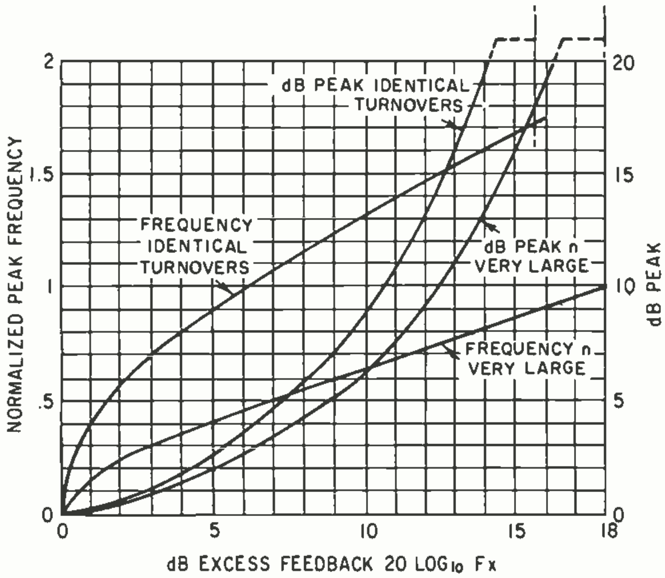

Figure 6. Chart showing variation limits for frequency and height of peak with circuits within boundary conditions of Figure 5. (From Crowhurst, N. H., High Fidelity Sound Engineering, *1961).*

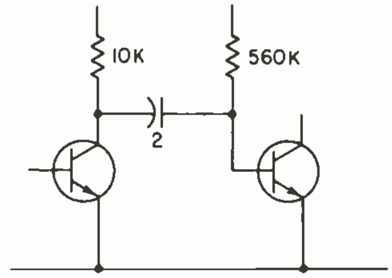

Figure 7. A form of coupling that can give trouble not predicted by the theory.

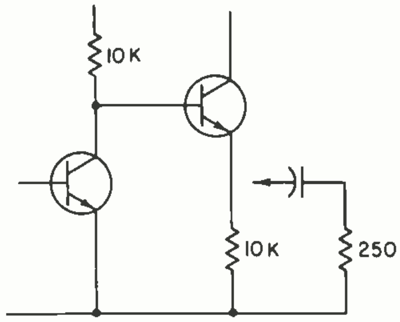

Figure 8. Another example: the well-known emitter follower.

impedance associated with that coupling capacitor is about 10 kΩ. A 2 μF capacitor will give a low roll-off of about 8 Hz. But now the signal level rises enough to swing this stage so that the second transistor just reaches cut-off. Two things happen.

All the while the transistor was conducting, the capacitor merely provided a.c. coupling. But as soon as it swings into its non-conducting region, it suddenly begins to act like a simple diode, using the 2 μF capacitor to build up a charge that sends it further into non-conduction.

And second, when it was conducting, that 8 Hz roll-off represented a time constant of 20 milliseconds. Now the relevant components are 2 μF and 560 kΩ, which represents a time constant of over a second. If you want the roll-off lower, as you probably would, the time constant is even longer.

That particular problem is relatively simple to correct. Just put, say a 10 kΩ resistor from base to ground. When the amplifier is amplifying, it will have almost no effect, paralleling the base input resistance of a few hundred ohms. But when it cuts off, it limits the time constant to, say 40 milliseconds.

That covers what happens at cut-off, in that instance. If you run into saturation, a similar thing happens; amplification, relatively suddenly, disappears. The parameters that are operative while amplification is present and on which the formulas in the earlier parts of this series were based, no longer apply.

What happens if you suddenly write zero for A, instead of whatever figure it has when amplification is operative? That cannot be expressed in a simple formula. You must look at the circuit and ask yourself what the condition is at each point around the circuit when this happens. You are referencing against time, not against frequency.

MULTI-PURPOSE FEEDBACK

For another example, take the well-known emitter follower. Suppose (FIGURE 8) you have a previous stage that uses a 10 kΩ collector resistor. You direct-couple this to an emitter follower with a beta of, say 40. By the simple rule for emitter followers, this will produce a reflected impedance of 250Ω.

Now, that emitter follower can do two things—change impedance and reduce distortion. We have already used its impedance-changing property. Assume its emitter resistor is another 10 kΩ. With a beta of 40 and a 10 kΩ emitter resistor, this puts its voltage amplification at 40, and its feedback factor at 41. If, as an ordinary amplifier stage, with no feedback, this transistor has a distortion of 8 per cent, then as an emitter follower with a 10 kΩ load, that ditsortion will be down to 0.2 per cent or slightly less. Since it is directly coupled, overall feedback could be used to knock distortion even lower.

But now, suppose you use the emitter follower so it can operate into the same source impedance it presents at the output, 250Ω. What happens now? Now its emitter load is not 10 kΩ, but 250Ω. Its gain is down from 40 to around 1. So $1 + A\beta$ is 2, instead of 41. This means that the distortion figure is about 4 per cent, instead of 0.2 per cent. Overall feedback may still knock it down, but it will be bigger, by about 20 times, than you expect.

At frequencies where phase shift comes into the picture, the distortion components, harmonic frequencies, will not be properly negative, so will not be reduced as much as calculated for other frequencies.

Another form of multi-purpose feedback, used in audio, is where you use it for gain control. Now you are not using audio feedback, as such, but are processing it, to get d.c., that controls gain. Of course, your d.c. is filtered. But that filtering must meet certain time-constant demands, to meet specifications.

This means that it cannot perfectly eliminate the audio. You think of it as d.c. feedback, but there is some audio feedback present as well. Never forget that.

This has been a sort of short course on feedback basics. I've tried to give you most essentials. How did you like it? Are there pieces you feel I missed? Let me know. ■

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• **William L. Starling** has been promoted to the post of western regional manager, professional products, for **Capitol Magnetic Products,** of Los Angeles. Mr. Starling who came to Capitol from **Data Packaging Corp.,** was formerly field service manager for Capitol.

• The newly created position of product manager, logging recorders at the **Scully/Metrotech Division** of the **Dictaphone Corporation** has been filled by **Leon A. Wortman.** Mr. Wortman has spent more than 20 years in the audio industry, associated with the **Ampex Corp.** and operating his own consulting firm.

• **Wally Heider Recording** of San Francisco has announced the appointment of **Gary Blohm** as general manager. Mr. Blohm was formerly West Coast manager of administration and recording operations for **Columbia Records** in Los Angeles. Among his new plans for the San Francisco facility are getting into radio drama production, more commercial recording, and service to students and community groups.

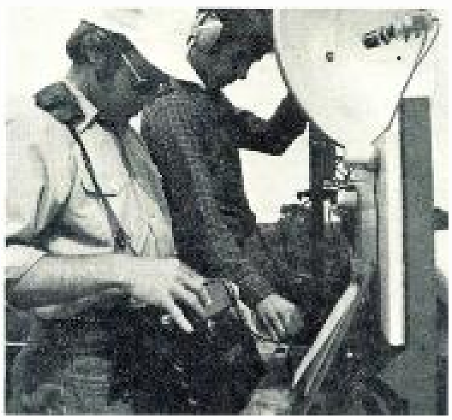
• The establishment of an international products division has been announced by the **Hoppmann Corporation.** The new division will offer standard components and accessory items to the communications market, portable a/v displays, personnel training cassettes, and intercom systems. The new division is focalized by **Horace Frenk** and **Ed Somerville.**

• Speedier delivery of **Bang & Olufsen's** audio products from Denmark to the U.S. is being effected through the use of jet air freight. Shipments which used to require three weeks to Chicago will now be delivered in one day.

• **Paul B. Ostergaard,** of Caldwell, New Jersey, has been elected president of the **National Council of Acoustical Consultants.** The Council is a nonprofit association representing acoustical consultants in the U.S. and several foreign countries. They are headquartered in Silver Springs, Maryland.

• A new acoustical engineering consulting firm, **DBH Acoustics,** has been formed in Portland, Oregon by **Lawrence G. Hopkins** and **Albert G. Duble, Jr.** The address is 10211 S.W. Barbur Blvd., Suite 209, Portland, Oregon 97219.

• **Jack R. Smith** has been elected to the position of Board Chairman at **Globe Communications** of Cleveland, Ohio. Mr. Smith was formerly a field engineer with the **FCC.**



• Project personnel of the **U.S. Fish and Wildlife Service** are shown preparing to locate and record howls and other vocal response of wild wolves on a **Uher** 4000 portable open-reel recorder. A radio-collared wolf is tracked through the use of a receiver which receives directional beeps. The point is to count the number of wolves, etc. in remote areas.

• **Jack K. Daniel** has been appointed director of marketing of the **Vega Division** of the **Cetec Corporation.** Mr. Daniel will have his headquarters in El Monte, California. Before joining Vega, he was with **Harris Communications.**

• **Warren & Hickey Sales Company** of Redwood City, California has been appointed by **University Sound** as their representatives for northern California and northern Nevada. Principals in the sales firm are **Don Warren** and **Bob Hickey.** University Sound is a line of the **Altec Corporation.**

• **John Snell,** formerly senior production engineer for the **ABC Radio Network,** has formed his own production and recording company. He will also serve as production/technical consultant to public relations firm **DWJ Associates, Inc.** of New York. Among Mr. Snell's assignments while at ABC were political conventions and elections, as well as several Gemini and Apollo space shots. His firm is located at 295 Madison Ave., New York City.

• New western representatives for **Analog & Digital Systems, Inc.** of Wilmington, Mass. have been appointed. **The Henry Joncas Company** of Seattle, Washington represents the northwestern region and the mountain states are now served by **MF Sales of Arvada,** Colorado.

• Offering a line of studio accessories in addition to studio design and construction services, **Windt Audio Engineering** is now operating from a new facility. The new office is at 13026 Saticoy St. (#4), N. Hollywood, California. **John Windt** is the owner.

• **Synapellas,** a series of quick synthesizer/a'capella jingles, are being offered by **WAY Audio Creations** of Buffalo, N.Y. The tapes feature **Roger Luther** playing the world's largest Moog synthesizer and are designed for any uptempo music format. Demo tapes are available from Way Creations, P.O. Box 21, Station B, Buffalo, N.Y. 14207.

• Of interest in applications requiring background music is the "Index Series" recently introduced by **MusiCues** of New York City, representing the **Chappell Background Music Library.** The series includes thirty-six 12-inch LPs and a compact catalogue, organized with one LP per subject matter. MusiCues is at 1156 Avenue of the Americas, New York, N.Y. 10036.

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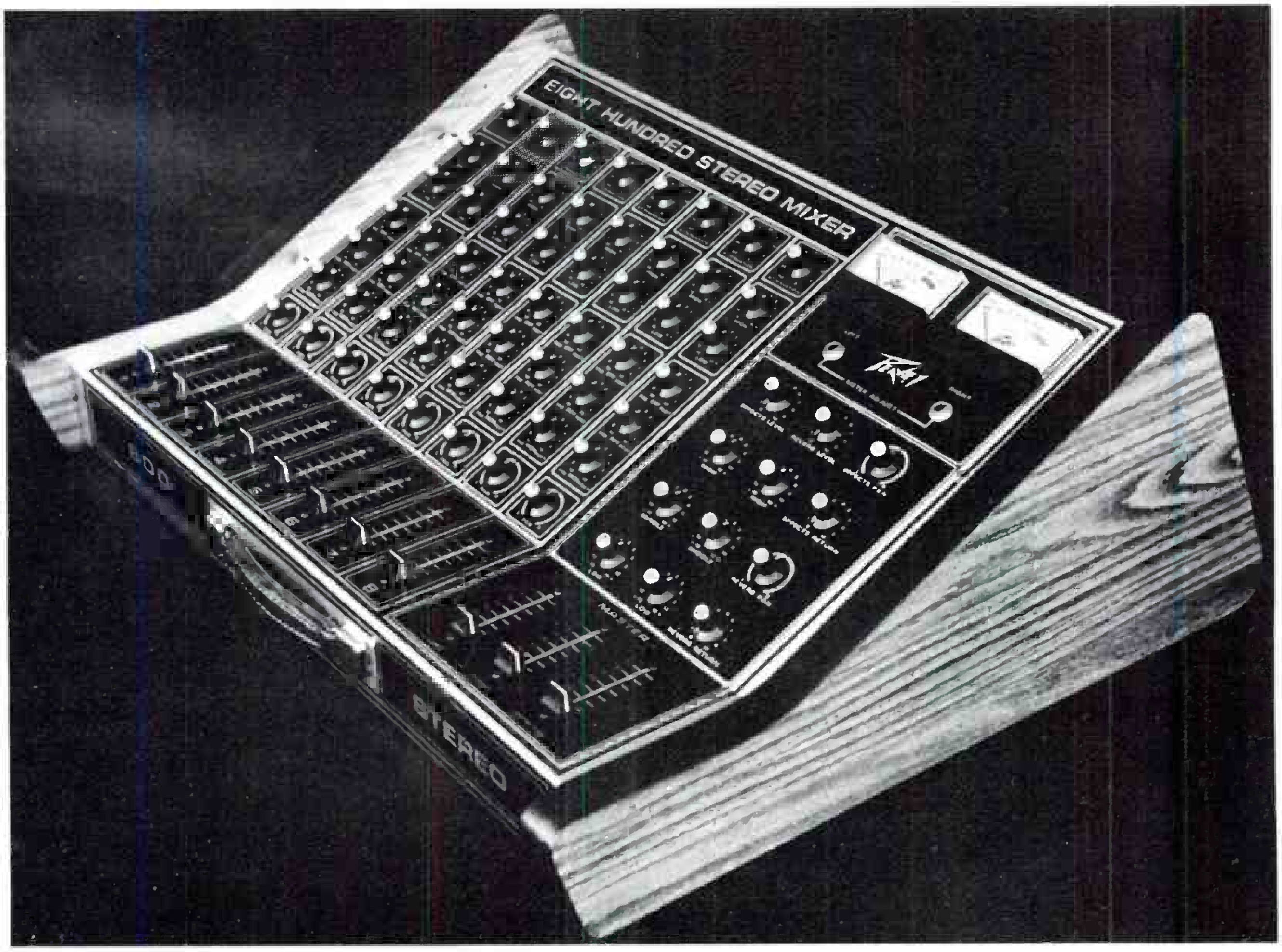

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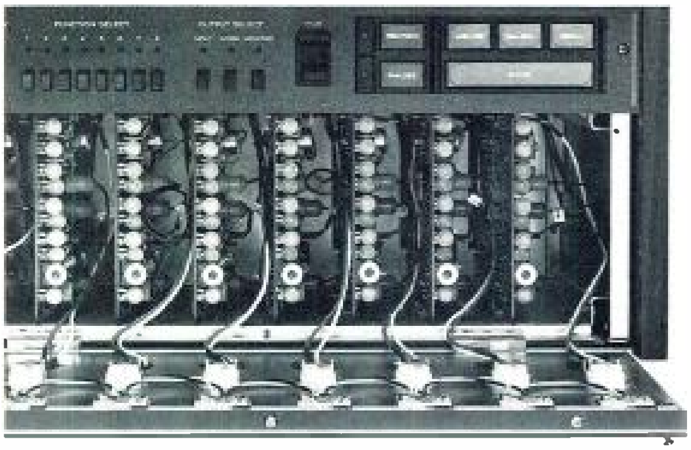

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