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JUNE 1977
VOLUME 8 – NUMBER 3

Recording engineer producer

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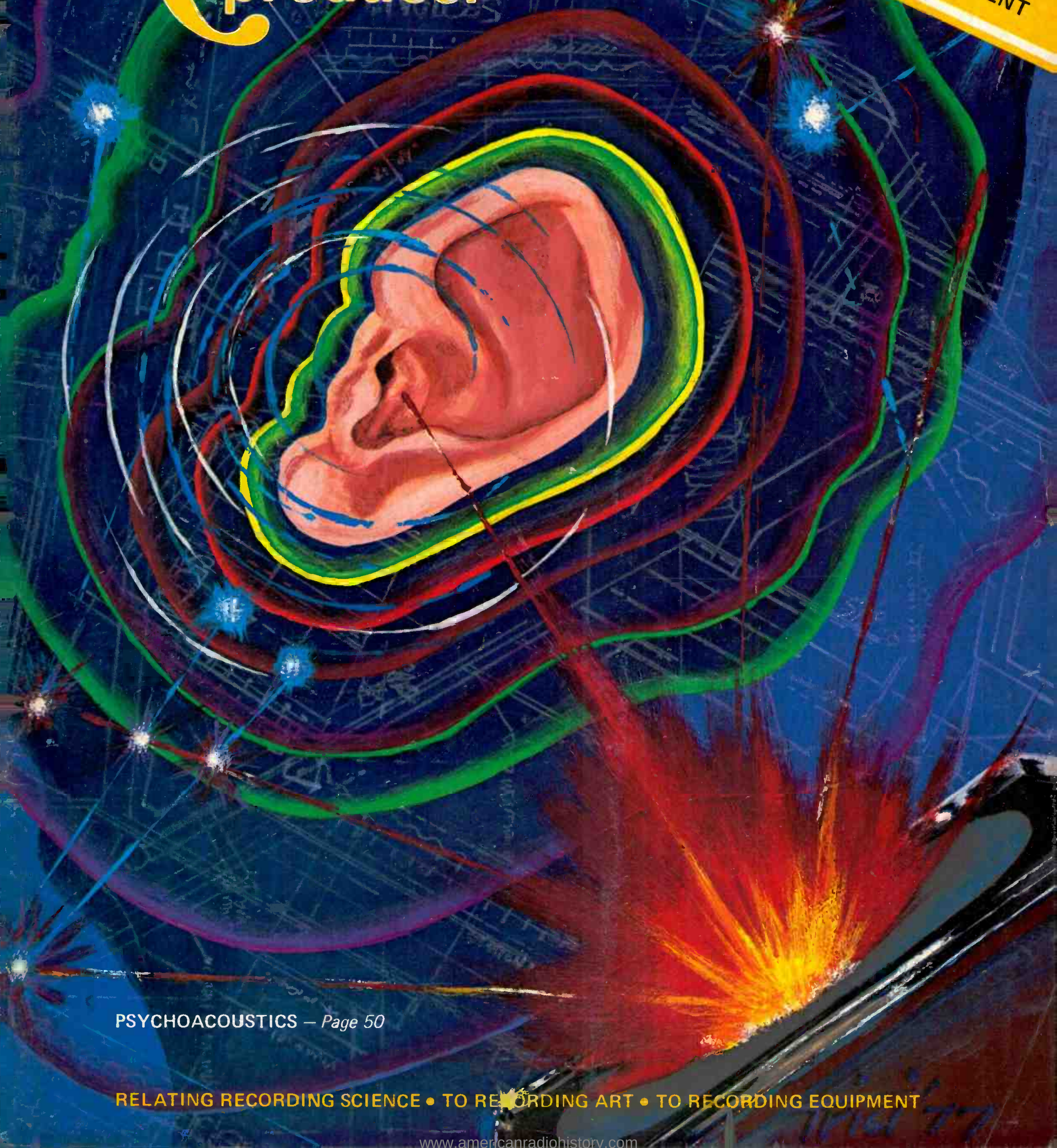

PSYCHOACOUSTICS – *Page 50*

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JUNE 1977
VOLUME 8 — NUMBER 3

—CONTENTS—

original cover painting: TRICI VENOLA

RECORDING engineer/producer

— the magazine to exclusively serve the Recording Studio market . . . all those whose work involves the recording of commercially marketable sound.

— the magazine produced to relate . . . Recording ART to Recording SCIENCE to Recording EQUIPMENT.

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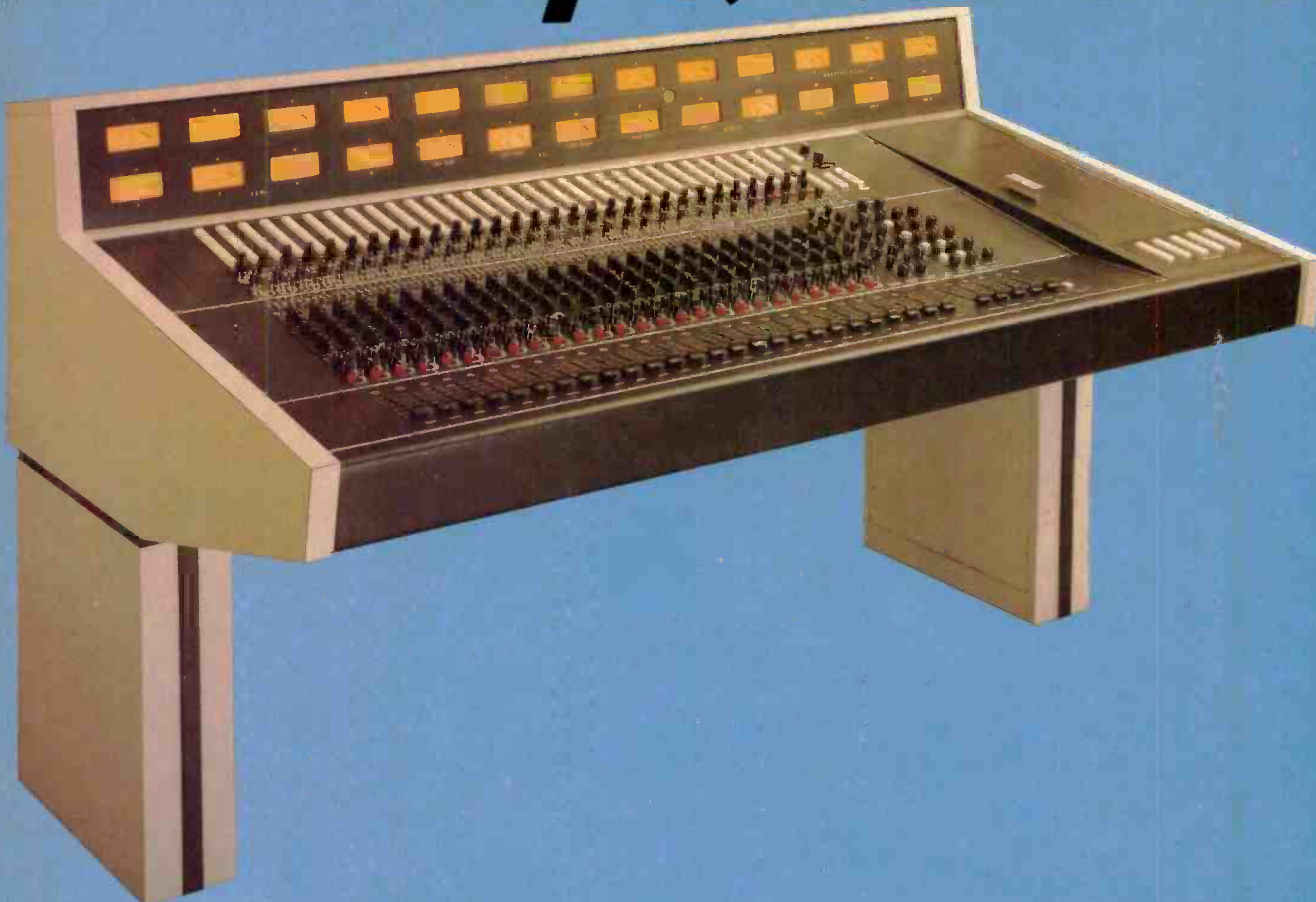
The Only Quiet One



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Letters and Late News

from: **PAUL E. ROLFES**
Chief Engineer
SOUNDCRAFTSMEN
SANTA ANA, CA.

Chris Foreman's article, *"Power: How Much Is Enough"*, in the April issue of R-e/p (page 81), was indeed a thorough discussion of the subject. Surely he couldn't have known when he was preparing his excellent article that Soundcraftsmen would be introducing a revolutionary new audio amplifier. Briefly, we would like to describe our new class H amplifier.

The class H amplifier is a new class of audio amplifier which is suitable for, and specifically designed for, the reproduction of hi fidelity sound. The first thing one would notice when looking at the differences in the class H amplifier is the fact that it has two positive and two negative power supplies. The ratio of the voltages of these power supplies is arranged such that the low voltage supply is two thirds of the hi voltage supply. In operation the amplifier appears to work exactly like a conventional class AB amplifier at low volume output, however as the signal level approaches the limit of the low voltage supplies a difference begins to become obvious.

Referring to Figure 1, the oscilloscope picture shows two horizontal lines which are the B+ and B- supplies to the output stages, and a 1 kHz sine wave operating within the limits of these supplies. As the output signal increases, one would expect that clipping would occur when the output level reaches the supply voltage level. However, as seen in Figure 2, the *Vari-Portional* circuit anticipates the sine wave's approach to the supply level and begins to increase the B+ to allow for additional head room. This process continues as required, until the Vari-Portional system has reached its limit and the sine wave has entered the opening made when the Vari-Portional system increased the B+ supply to provide more head room to the amplifier. If the output level continues to increase, clipping finally will occur against the high voltage supply, as seen in Figure 4.

The obvious advantage of such a system is that the amplifier is operating at a lower voltage most of the time. This lower voltage operation saves energy because it substantially reduces the dissipation of the amplifier, since the dissipation of the power output stage is directly proportional to the voltage applied across the output transistors. It should be noted, however, that there is an energy saving at all times even under high power sine wave conditions. Referring to Figures 3 and 4 it can be seen that although the hi voltage supply is being turned on to its maximum, it is only on during that period of time when it is required. It is still off for a substantial portion of the sine wave, consequently the amplifier is operating on the low voltage – (the more efficient supply) – during this period of time.

Going back and reviewing the sequence

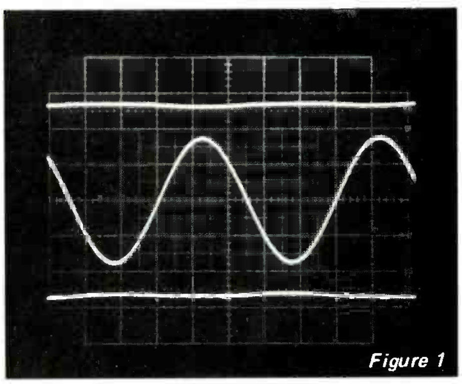

Figure 1

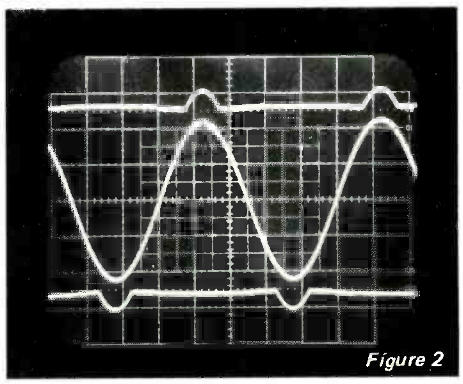

Figure 2

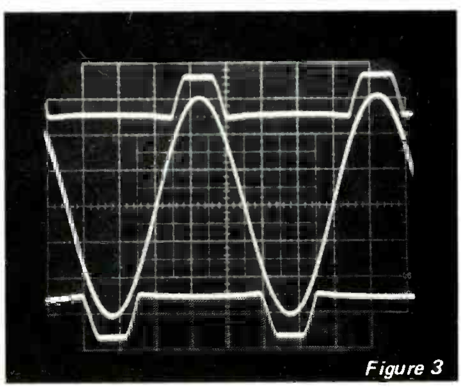

Figure 3

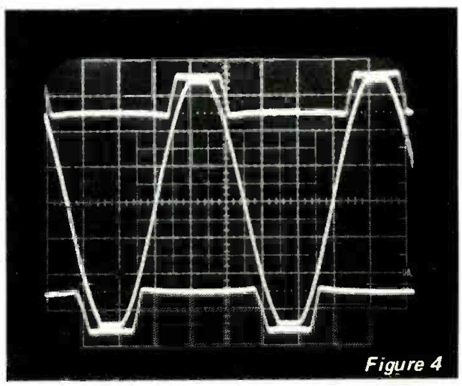

Figure 4

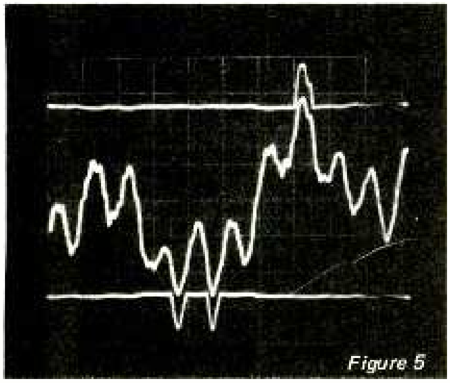
Figure 5

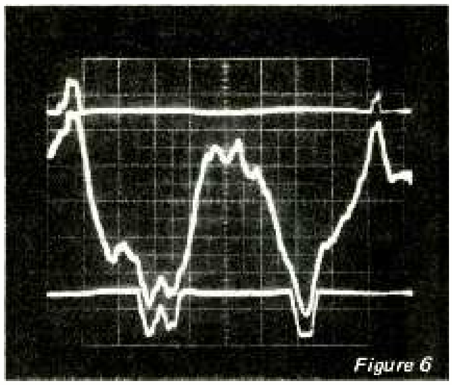
Figure 6

of these pictures again, it can be noted in Figure 1 that the hi voltage supply is not required and therefore not used. In Figure 2, the hi voltage supply is actually not yet required because of the selection of the output voltage, however the Vari-Portional logic circuitry has detected a rising wave shape, is anticipating the fact that the supply *may* become required, and therefore is beginning to turn the supply on *in advance* of the sine wave actually reaching the supply level. In Figure 3 it can be noted that the slope or rate of rise of the upper wave shape (which shows the positive supply turning on) is *greater* than the slope of the sine wave that is entering into the upper supply. Because of this fact, the sine wave can never "catch up" with the upper supply, since the "turn on" gain of the upper supply is *greater* than the gain of the amplifier. It should also be noted that the inherent slew rate of the upper supply is *greater* than the slew rate of the amplifier, which is approximately *50 volts* per micro-second. Therefore, regardless of wave shape (even a hi frequency square wave) the Vari-Portional supply logic is fast enough to anticipate the rising wave shape and turn the supply on with a gain and slew rate that are *higher* than the amplifier and therefore move out of the way of the oncoming output signal.

Referring now to Figure 5 and 6, you can see actual oscilloscope pictures of loud rock music and the complex wave shapes involved in music of this type. In Figures 5 and 6, the signal is penetrating through the positive supply voltage and is entering into the area of the Vari-Portional control system. In Figure 6 note that, toward the left hand side, the B- supply began to turn on *before* it was actually called upon for use, by *anticipating* the rising wave shape. Toward the center you can see how the B+ supply was called into service as required.

From the above description of the operation it can readily be seen that the amplifier using the Vari-Portional control system, or class H, achieves a substantial savings in dissipation or heat loss on the output transistors.

The major advantage in using this method over other methods for reducing the dissipation on the output transistors is that there is no *switching* or *changing of signal paths* within the basic amplifier itself. All of the controls for increased power requirements act *only* within the power supply and therefore are outside of the feedback loop and have *no effect* on the distortion, stability or slew rate of the basic amplifier. All of the advantages of the class H circuitry have been included in the very conservatively rated Soundcraftsmen Model MA 5002 Power Amplifier which is rated at 250 watts per channel into 8 ohms both channels driven.

Editor's note:

An additional product review appears in the New Products Section.

STUDIO ON ICE

from: DEBORAH & TED ROTHSTEIN*
WOODSTOCK, NY

. . . Readers of R-e/p might be interested in our visit to a state-of-the-art, 24-track studio located in, surely, one of the least likely places on earth – ICELAND.

We arrived at Kennedy International Airport not knowing what was in store for us. After all, most Americans, if they see Iceland at all, pass through en route to or from Europe; few, if any, stay to see the place and the people.

We were intrigued. Iceland, "the land of ice", home of glaciers, the first parliament and ancient land of the Viking kings now had its own 24-track recording studio. But don't let the name cool you off. For one thing, coming from the record cold "winter of '77" we were ready for anything, and as it turned out, Iceland was indeed a welcome change. Well-passed its "low season", (around Christmas when daylight is negligible), we flew into the sunrise at Keflavik (the U.S.-N.A.T.O. base which serves as Iceland's international airport) to comfortably above-freezing weather. Although this *is* unseasonably warm, the average winter in Reykjavik (capital city) is warmer than we are accustomed to in our hometown of Woodstock, New York. The climate is actually similar to that on the northwest U.S. coast.

We flew over with Sigurjon Sighvatsson, the studio's manager, and Jonas Jonsson, a studio engineer and Icelandic TV personality – two young hip guys (among the founders of the studio in Jafnarfirdi, Iceland) who were escorting us and some equipment to their studio, and architect John Storyk, from Sugarloaf View, Inc., in New York. We had embarked on a journey to this strange island nation, timing our travels to do a little sightseeing and take in the Paris AES show as well.

Geologically and sociologically Iceland is a country still in the making. The people enjoy working and work hard, yet also know the meaning of a good time. With the studio management we hit the nightlife and fine restaurants and saw some of the unique countryside.

The brochures say it is a country of contrasts, and that is quite true. The name "Iceland" depicts only a small portion of what the island is like and gives no inkling of what it has to offer. The land of glaciers and geysers is also the land of the midnight sun and the northern lights, hot springs, and bright flora, abundant fishing and the open range, and is one of the chess capitals of the world. As they have abundant geo-thermal heat, (you literally stick a pipe in the ground and have instant and unlimited heat and hot water) there is no worry of fuel shortages. The people of Iceland enjoy a comfortable standard of living, essentially all being middle class. There are still only 60 miles of paved road on this island country the size of Kentucky, and people still opt to ride horses instead of automobiles.

Then why a recording studio on Iceland? As is not uncommon, it was begun by a few musicians who initially conceived of the studio to provide Icelandic musicians with the fine quality facilities, the lack of which had caused them to go elsewhere in the past (primarily England). Like the rest of the world, the fruits of the music business are abundant in Iceland. The level of audio consciousness, high. In addition to Icelandic music, the people listen to all the top rock, pop, and jazz stars of the U.S. and Europe. The statistics attest to this fact; over 300,000 local record products and over one million foreign records and tapes were sold in one year in a country with a total population of 200,000! The decision to build a studio was only natural.

As many artists have found, the location and atmosphere of the recording studio is as much a part of their end product as the differences among musicians and the music itself. Hljodriti Studio in Iceland can give a travelling musician a quiet place to get down and work, to stop over from a busy tour in Europe or the States and recover, relax, and work out some ideas, lay down a few tunes away from the hectic schedules at home.

The name Hljodriti (pronounced he-olth-writ-te) is the Icelandic word for the pioneer Edison cylinder phonograph. With that in mind, the studio represents just how far the audio industry has come in this century. In the brief history the studio has had, they have moved quickly

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The MCI equipped control room.

and effectively from an adequate 8-track installation to full 24-track state-of-the-art performance and design, placing them easily on a par with their American and European counterparts.

Hljodriti was conceived in December 1974 and opened as an 8-track facility in April 1975. It was re-designed and up-graded to a full 24-track studio last June. The principals are all musicians or former musicians, quite sophisticated and professional in their approach to music and the recording field, and they all speak and understand English very well. Among their personnel, the Chief Engineer, Tony Cook, is from England, with a background in the British recording industry. The maintenance engineer, Baldur Sigurdsson, is a European-trained Icelander (and having spent some time with him in the studio, we could see that he was well-equipped to deal with problems that might arise). In appearance the studio is meticulous and modern, with all components carefully matched. As you can surmise, we were pleasantly surprised to find a recording studio of this caliber in Iceland.

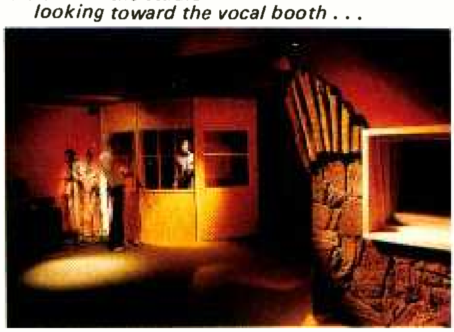
Getting down to basics, in their bevy of equipment, they have the standard array of microphones from these companies: Neuman, AKG, Sennheiser, Beyer, Sony, Electro-Voice and Shure. They use Dolby noise reduction for multi-track and two-track tape machines. The outboard gear is also state-of-the-art equipment, including limiters from Eventide and dbx, ROR Parametric and Pultec equalizers; high quality echo is provided by an EMT 140 and a Master Room echo chamber. They have a harmonizer from Eventide, the fully-loaded model with keyboard and all other options. Also from Eventide is the flanger (and two omnipressors). All of this is linked together by an MCI 428, 28-in, 24-out console, feeding an MCI 24-track tape machine. The Scully 280 is used to mix down 24 to 2 and the Revox A700 provides the copies. Cassette copies can also be made. They are monitoring with JBL 4320's which were given flush installation for maximum bass response, and are equipped with passive 1/3rd octave equalizers for minimum noise and maximum reliability. A B&K sound level meter with 4165 microphone, and an H.P. 8050A real time spectrum analyzer were used to adjust the equalization of the control room monitors. They also have mid-sized bookshelf speakers and small Auratones.

The acoustics (treatment and shape of the room) is what can set one 24-track studio apart from another, and in that respect Hljodriti ranks with the best, having been designed by John Storyk (Electric Lady, Atlantic, Leon Russel's Studio and Sigma-New York). He directed these and other studio projects as chief architect with Sugarloaf View, Inc., the New York based architectural and acoustical design and construction firm.

According to John, the control room is a semi-compression, semi-trap design. We measured the reverberation time with an Acoustilog 232 reverberation timer and found it to be very close to the average living room with good low frequency absorption. (See graph.) The B.B.C., which has done the most research in this field, has determined that control room reverberation time should be similar to that in the average listening room (living room).

The control room has a refreshing environment. There is extensive use of wood and lava rock (indigenous). It is extremely large and comfortable measuring about 400 square feet (37 square meters) and has a seating capacity of 15. There is carpeting, wall-to-wall. The back wall is a convex, horizontal broad-band wood slat resonator. The power switches are cleverly concealed behind the wood slats. All surfaces are durable and are done with local materials, except the wood, which is clear fir imported from Norway. All the woodwork and detailing is precise and clean.

view from the studio looking toward the vocal booth . . .



The basic audio layout is essentially American, therefore providing a very comfortable recording system for American artists/engineers/producers, however, there is an interesting blending of European and American designs and concepts. The furnishings and the architecture are a blend of the two cultures. All the light fixtures and hardware are Swedish, and are exquisitely contoured, feel luxurious and work beautifully. The voltage is European, but they have provided high-power transformers so that 115 volt U.S. equipment can be operated on 230 volt lines.

The studio is equipped with a Martin Guitar D-43, Fender Jazz Bass, Ampeg B-15 and Fender Twin amps, and a new Yamaha Grand piano, Model S-7. Hljodriti has recorded pop, rock 'n roll and jazz musicians, and has done classical sessions as well. For the artist, there is in Iceland, a high availability of excellent studio musicians who are European-trained and cover the whole gamut of instrumentation.

In the studio, as in the control room, there is a large use of natural materials (wood and lava), giving the physical space a texturally unique quality. The design of the studio is catered to a variety of music needs and includes a live area and a dead area. The enclosed vocal and drum booths have been carefully, ergo, dynamically placed in the studio.

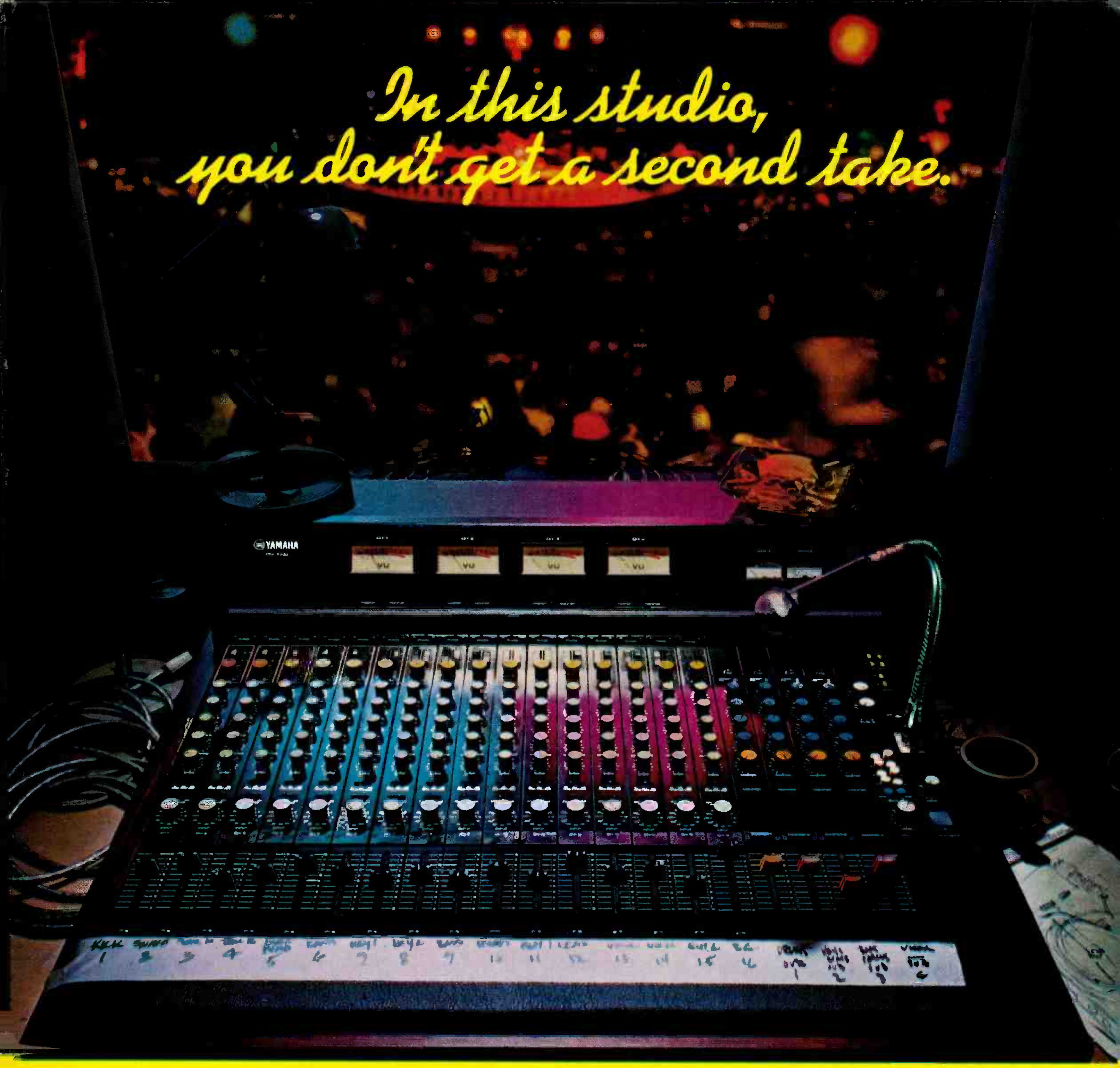
In making sound transfer measurements we found that the isolation between rooms is better than 48 dB even at 125 cycles. Reverberation quality in the studio is excellent. As the graph indicates (see graph), maximum variation between octave center frequency measurements is never more than .1 seconds.

There is substantial enough difference in reverb times between the live and dead areas to create all the necessary recording environments.

One of the major causes of ambient noise in a recording studio can be the air conditioning system. This studio doesn't need one; the reason is that the ventilation is taken care of by just opening the window, which they can do because they are in a remote area, even though they are only 10 minutes from Reykjavik. Consequently, Hljodriti has an exceptionally low ambient noise level. The studio is very quiet (N.C. <15). In fact, when Ted and John initially measured it, they thought the sound level meter was broken because the readings were so low.

The aesthetic of the studio is again, like the control room, one of space and polish, enriched with texture and trim styling; the room measures 950 square feet (88 sq. meters). Hljodriti has as much dynamic range and quiet as the $180 per hour American studio. The cost of recording at Hljodriti is $60 per hour, which includes studio time, air flight and hotel. The studio is 100% musician oriented. There is virtually no advertising in Iceland, due to the way in which the media is handled (similar to B.B.C.) so there are no com-

Looking toward the control room showing the wall and ceiling acoustical treatment.



When you perform in front of a live audience, you put everything on the line.

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mercial considerations subtracting from that orientation.

A look to the future of Hljodriti envisions a studio that has tapped the interest shown in the international market. They are also currently in the process of expanding their facilities to include a large rehearsal room, also designed by Sugarloaf, and a tape duplicating facility. Unquestionably, these enterprising gentlemen have injected a progressive element into the Icelandic culture and raised the level of recording substantially. They are not finished here, however, and intend to maintain the standards they have set as the industry continues to grow.

biographical data:

DEBORAH and TED ROTHSTEIN

Ted is an audio engineer with a background in electrical engineering, audio and P.A. maintenance and design. He is currently chief engineer of Bearsville Sound, Bearsville, N.Y., and vice president of ROR Audio Research, Inc., New York City.

Deborah is a freelance writer, wife and mother of 3 year old Tara Rothstein.

ERRATA

In our last issue, Leonard Kovner was named as having done some engineering on Boz Scaggs' "Silk Degrees" album. Though he has worked with Boz in the past, and parts of the "Silk Degrees" album were done at Davlen, Kovner did no engineering on the project. The engineer was Tom Perry.

FILMWAYS/HEIDER RECORDING ANNOUNCES MOVES

Filmways, Inc., has announced that it has entered into an agreement with RCA Records to acquire and operate the former RCA Recording Studio facilities in Hollywood. The studios will be operated along with the Wally Heider Recording Studios, already owned by Filmways, under the new name "Filmways/Heider Recording", and it is claimed will constitute the country's largest independent complex of recording studios and remote facilities for both the recording and broadcasting industries. Filmways/Heider will be able to provide complete recording, mixdown, mastering, editing, and production services through the lacquer disc stage.

In conjunction with this expansion Filmways also announced a reorganization of all its audio related businesses into a new Filmways Audio Services Group with Laurence Estrin as President. The Group will now consist of Filmways/Heider Recording, whose President is Ron Trowbridge; Filmways Radio, which provides automated programming and related services, with Gary Standard as President; and Filmways Equipment Services (formerly Filmways Audio Services) which provides sound reinforcement services to a wide variety of clients. Laurence Estrin remains President of Filmways Equipment Services.

PROF. CYRIL HARRIS GARNERS MAKER OF THE MIKE AWARD

New York's Avery Fisher Hall was appropriate setting for presentation of the coveted Maker of the Microphone Award to Columbia University's professor of architecture and engineering, Dr. Cyril M. Harris. Given annually in memory of microphone and disc record inventor Emile Berliner, to commemorate an outstanding contribution to the world of sound, Professor Harris received the award in recognition of his superior acoustic design of Lincoln Center's Avery Fisher Hall, previously the root of much concertgoer consternation. Presenting the trophy is Oliver Berliner, right, grandson of the inventor. M.I.T. graduate Harris is a member of the National Academy of Engineering, Fellow and Past President of the Acoustical Society, and AES Honorary Member. He is responsible for the acoustic design of Washington's Kennedy Center and the Metropolitan Opera House of New York among other notable edifices.

UREI IN NEW HEADQUARTERS

United Recording Electronics Industries (UREI), a major producer of professional audio measurement and signal processing products, has relocated its entire manufacturing, sales and administrative offices to a larger, more modern and efficient facility recently purchased in Sun Valley, California.

Located at 8460 San Fernando Road, Sun Valley, California 91352, the new plant has 21,000 sq. ft. under roof and an additional acre adjacent for expansion. Situated just off of the Golden State Freeway in the San Fernando Valley area of metropolitan Los Angeles, UREI is now easily accessable to rail and truck service and nearby Hollywood Burbank Airport.

"The expanded facility will enable us to fabricate in house many components which previously were purchased. This will provide better inventory control and production efficiency. Additional manufacturing and storage space will also allow us to maintain larger inventories of finished products and to accommodate new items now under development, including a line of super monitoring loudspeaker systems now being introduced. UREI also plans to acquire and develop additional products for the commercial sound and broadcast markets." So states Bud Morris, UREI Executive Vice President and General Manager.

UREI's Telex number remains the same at 65-1389 (UREI SNVY) however, the new telephone number is (213) 767-1000.

UNI-SYNC BECOMES UNIT OF BSR (U.S.A.), MOVES TO LARGER QUARTERS

Designers and manufacturers of sound reinforcement equipment, Uni-Sync has been acquired as a subsidiary of BSR (U.S.A.), Ltd., according to John Hollands, President, BSR, and has expanded into new facilities located at 742 Hampshire Road., Westlake Village, California. Telephone (805) 497-0766.

Uni-Sync will continue to be headed by former owner Michael Ragsdale, President. Larry Jaffe, an industry technical marketing consultant, has been appointed Marketing Manager.

The company will continue to supply commercial sound reinforcement equipment with principal emphasis on live music mixing systems. Marketed under the Trouper Series trademark, the current line consists of four sound reinforcement consoles, as well as accessories, ranging in suggested retail value from $750 to $3,000. The newest model, Trouper I, was recently successfully introduced, and has a suggested retail value of $750, making the unit especially applicable for small music group sound reinforcement systems.

continued on page 82

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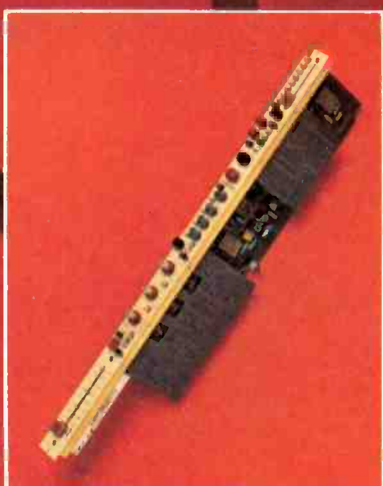

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We know that buying a large multi-track console is no small decision. For most professionals, it's one of the largest investments you'll make — a decision that you'll have to live with for years.

There are a lot of companies making consoles. Many perform adequately. Others are compromises. Few have all of the features and performance at a reasonable price. So, what are we leading up to? A simple statement of fact that you should consider seriously if you're really interested in an outstanding console system: Quad/Eight has an enviable reputation for quality and reliability. It's something we've worked at for over 10 years. We've also had a reputation for building the industry's most expensive systems too. Now, relax. Our new modular series consoles look expensive. Truth is, they're priced right in the same category as our best competition. In addition to having the best human engineering for operational ease, they're loaded with more features and performance:

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*The Coronado, 40 Input/24 Output equipped with Compumix III available in October, 1977.

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TV audio:

Text and photos by Howard Cummings

JOE RALSTON originally started as a photography buff in upstate New York, but shortly thereafter began audio training at a small TV and radio station. Upon moving to the West Coast in 1965, he began work in the film department at NBC-Burbank after "being without a job for 6 to 8 months."

The first real audio work he did at Burbank was with the KNBC local presentations, such as news, public service productions, and the LOHMAN & BARKLEY comedy show. After getting the itch to move out of local production and into national coverage, he replaced a retiring mixer on the RED SKELTON SHOW. Audio for the DEAN MARTIN HOUR, LAUGH-IN, and BOBBY DARIN followed, along with DOUG HENNINGS MAGIC SHOWS and THE MIDNIGHT SPECIAL.

Howard Cummings: Could you contrast recording studio equipment with TV equipment?

JOE RALSTON: In the recording studio, they try to take the shortest distance from the mike to the tape head. In television it's a different story. From the mike to the board is one thing. But from that board out it's got to go to a lot of other places — it's got to go to video tape, but it *also* has to go to dressing rooms, it has to go back on the stage for P.A., it has to go to the orchestra, it has to be split and go to a client's booth.* So now you have all these feeds and because someone wants to hear one thing and not another, you've got to start picking it off at different points. You can't send a final mix all over — you've got to give them a mix-minus; the announcer wants to hear everything, but not himself, the people on the stage want to hear everything but not audience reaction machines, and we don't want it going into the house.

e boards themselves have to be able k up the signal, whatever it may be, at different points along the line, and feed it someplace. To get a certain level to feed it, we have to pad it, boost it back up, and right away our signal-to-noise is not what it is in the recording industry.

Howard Cummings: Around 50?
Joe Ralston: It depends. With THE MIDNIGHT SPECIAL – we're working with four stages in a circle – and at least 3 of the 4 stages have big neon signs with the name of the group behind them. So you've got an AC problem and a hum problem. You've got groups that have just come off the road with amps buzzing – but who knows that in a stadium? When they go into a recording studio nothing moves until that buzz gets taken care of. A lot of things can be done in a recording studio afterwards, but television post-production is limited.

Howard Cummings: Do you notice a signal-to-noise loss by the time it goes from the stage to electronic editing to final broadcast . . . something that you can actually discern on TV?
Joe Ralston: (pauses) Probably not. If I listen to the sound I get when I record, and then listen to it at home, I know there's a difference. But to analyze it and to figure out where it's coming from, it's difficult. It's cumulative. It's from 5-6 generations.

Howard Cummings: Literally *5 or 6?*
Joe Ralston: Sure. It happens.

Howard Cummings: So you think you start with 50 (dB S/N)?
Joe Ralston: Depending on the studio and the equipment you're working with.

HC: Then when it's down 5 or 6 generations for final broadcast you might be in the 35-40 range?
JR: I couldn't really put a number on it. To put a number on it would be tough because there's so many variations. The transfers back and forth, the electronic editing, the amount of post-production, the stage or electrical problems with the lights, then were're getting into a *lot* of noise.

I've seen some of these game shows that have . . . just thousands and thousands and thousands of feet of wire and peanut lights and flashing and chasers. You open a mike on a stage like that, you can just hear a "bed" of hum and buzz, and a lot of it is in the induction you get. There's just no way out of it.

HC: What about your mike inventory?
JR: We have condenser mikes, but for television shows we have convenience problems with AC condensers and their power supplies – so we went to the batteries.

We have almost all of the Electro-Voice line, almost all of the Shure line. We try a lot of mikes.

HC: So do you think it would correspond more to a P.A. situation?
JR: Probably more-so because we're dealing more with that.

Every group that comes in has to hear themselves with that stage foldback. We could use cans but we have a picture problem and we have to watch directionality on mikes that will give us a P.A. problem.

With mikes, we have to keep in mind that they're going to be used on MIDNIGHT SPECIAL one day and on a game show another day. We also have to keep in mind the picture. A U87 is a great microphone, but you put a singer on-camera with it and all you see is a microphone in front of their face. The picture is a big consideration, but the sound difference is not as noticeable over a TV speaker.

HC: It surprises me that they use 421's for HOLLYWOOD SQUARES.
JR: We probably tried about 4 or 5 different mikes and speakers before coming on the 421-speaker combination. Each one of those squares has a speaker right over it so they can all hear each other. That was a P.A. thing. Also you have to keep those 9 mikes open at once because you have 9 stars that are going to say something funny and not on cue.

HC: So there's no fader manipulation or gating of unused mikes?
JR: You're starting off with 9 right away and you've got to have them open because it's an ad-lib sort of thing. There's a pattern but if you miss a line from Paul Lynde, that's the name of the game – the joke, the laugh. If you gate things, you have another problem.

We have a studio sound, an ambience, and if you're using two booms and you've got them open and there's a joke told, the audience laughs. You're getting audience mikes with the laugh and applause, but you're getting a lot of leak in the boom and that adds to this full sound. If you close the boom it starts to sound very dry and you'll start to hear a lot of little "holes" in the ambient sound.

When we did WHEEL OF FORTUNE, we had a little problem with the giant roulette wheel and that damn thing has the loudest clapper in the world and they spin it every single time. If you open all the mikes and leave them all open, that's all you hear because you're using ECM 50's lavalieres.

A lot of things can be done, but we have a time limit to correct a lot of things. The idea is to get the show on the air. If you have a lot of time to really polish it, OK. The act will come in and be prepared and you'll be prepared or you're not going to do the show.

One of my biggest pet peeves is: I think more and more it's being dictated to the engineer how to mike the show.

HC: By whom . . . circumstances?
JR: Production people generally. They don't want their MC to have to have a cord so we have to go to RF. RF creates a lot of problems, but we get by.

HC: A lot of potential *problems or a lot of* definite *problems?*
JR: A lot of potential problems. There *are* some definite problems depending on how the set is built, where the cameras

*Client's booth: Conference room where advertisers, producers, managers, etc., can watch and discuss a show. Separate from audio control room.

BRH90 RADIAL HORNS

This is the BRH90, a ninety degree radial horn for two inch and one and three-eighth inch compression drivers. Its an easy name to remember —Big Radial Horn, 90 degrees— very simple. But there's nothing simple about our design or construction.

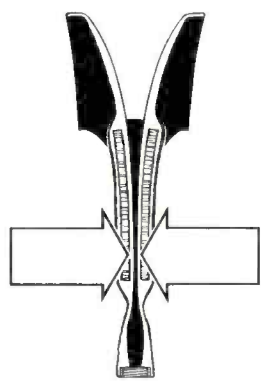
Community horns are made of fiberglass. Not sprayed-up cheaply-made fiberglass, but fiberglass hand-laminated by experts and constructed to our exact requirements for maximum acoustical accuracy, resonance-free rigidity and unparalleled strength. Our horns are absolutely weatherproof. They will never corrode or rust out. Nor are they ever likely to break. Not now and not forty years from now.

But the real mark of Community radial isn't apparent unless you cut it in half as we have done here. If you look closely at the cutaway, you will see that the corridor in the horn just past the throat gets extremely narrow before it flares out. That pinch in the horn is absolutely necessary to any mathmatically and acoustically correct radial horn, whether it is metal or fiberglass, and only Community does it the way it should be done. And not only do we do it right, we do it cheaper.

We also make two 90° radials for use with one inch and screw-on drivers —the RH90 and the SRH90. Both have the same flare rate, but the SRH90 (Small Radial Horn) has a smaller mouth, and is usually used as the high end in three way systems.

Talk to your Community distributor. Even though he also sells our competitor's products, he'll probably tell you that there is nothing available today that equals a Community horn.

He's right.



	BRH90	RH90	SRH90
Flare Rate	240Hz	345Hz	345Hz
Operating Range	500Hz up	600Hz up	1,000Hz up
Size: H.	11 1/8"	12 1/4"	6 1/2"
W.	33 5/8"	30 3/8"	24 3/4"
D.	21"	20 1/2"	18 3/4"
Weight	25 LB.	20 LB.	12 LB.
Finish	Black, High Gloss	"	"

	BRH90 KHz	BRH90 -3dB	BRH90 -6dB	RH90 KHz	RH90 -3dB	RH90 -6dB	SRH90 KHz	SRH90 -3dB	SRH90 -6dB
Horizontal Dispersion	.6	85	95	.6	80	90	1.2	95	100
	2	90	90	2	90	100	3	90	95
	10	80	90	10	85	90	10	85	100
Vertical Dispersion	.6	50	90	6	55	100	1.2	50	70
	2	35	50	2	35	50	3	40	65
	10	20	35	10	20	35	10	20	30

Community

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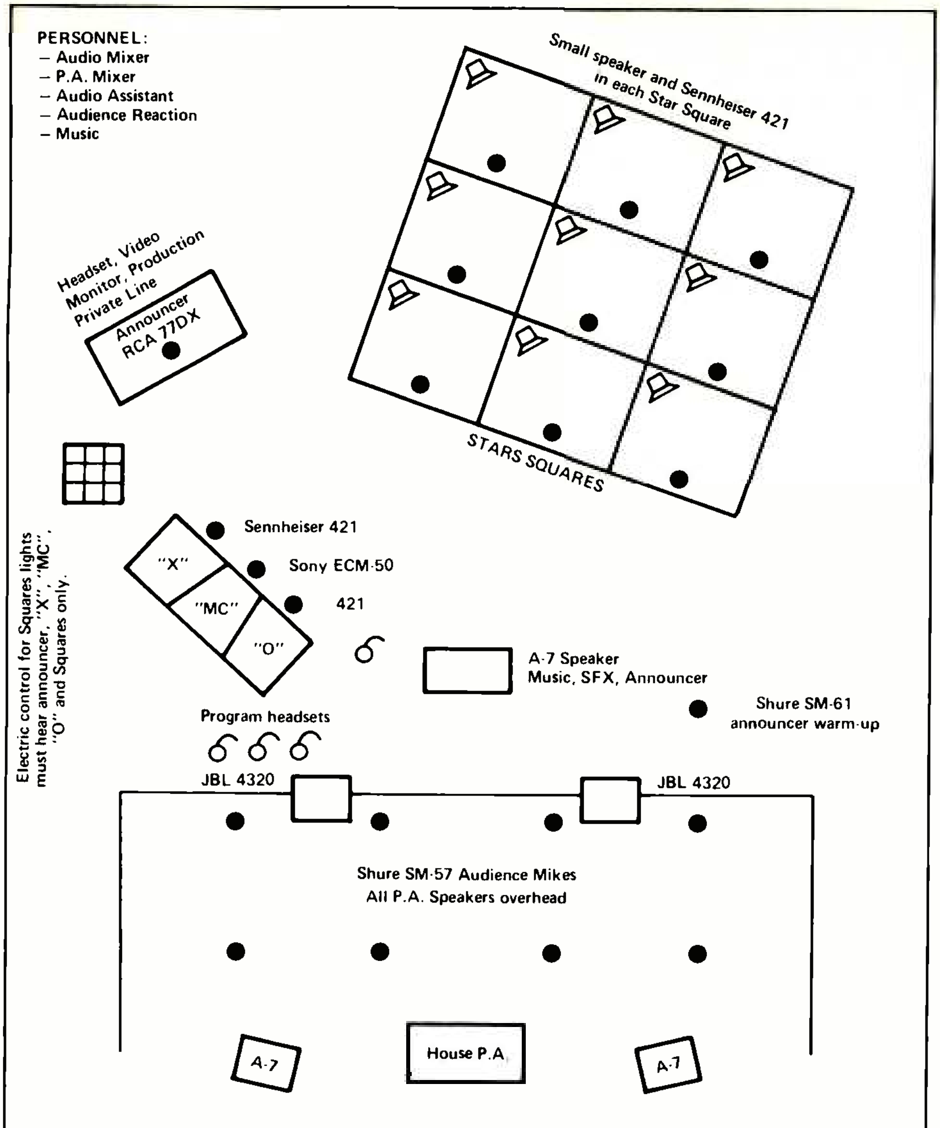

HOLLYWOOD SQUARES – Engineer: Joe Ralston – KNBC – Burbank

are located, where you'e going to have to put your receiving antenna, but people are being educated. They go to Vegas and watch TV and see RF. What they don't realize is that a game show may have been stopped seven times because of the RF. It looked great when they saw it, but it was a big problem getting there. Some potential problems would be: you go into live TV and they want to use RF for awards shows. What if it goes out, and you lose it, or in the middle of a big song you hear a taxi? I'm not against RF's *if they're used right*. It's the old story: the right tool for the right job. It's a challenge to mike something without having it show in the shot, but still you have to compromise, in most situations, on your sound.

If a producer wants a number to be done on a boom – a big up-tempo live orchestra song where you might have people with weak voices – the producer may *insist* that it must be done on a boom because he doesn't want to see microphones. I don't feel we are fooling anybody. If the public is sitting at home watching, they know there's a microphone someplace. So you see it, so what! I know there's arguments against this but that's my feeling. I'd like to do it nice and clean, looking clean, but if you want good sound, you have to work at it and you have to have the right tools to do the best job. If you have the wrong ones, you're hindered.

HC: What are your feelings on shotguns?
JR: We use modified shotguns.

HC: You say "modified".
JR: They're not *really* shotgun mikes.

HC: Wider spread?
JR: Yeah. People say "shotgun mikes". I can remember hearing about shotgun mikes and you could point it and hear something 30' away. Well, OK, but what did you hear? You heard a lot of garbage and a lot of ambience. We use shotgun mikes on the booms and right away everyone says, "Why is the mike so tight? It's a shotgun mike, you don't have to get it in so tight." (chuckles) I say, "It's a misnomer." But we use them a lot, the Sennheiser 805's, 415's, 815's.

HC: And yet I notice on the CARSON show they use . . .
JR: . . . a CS15 Electro-Voice, a good condenser microphone.

HC: And it's not a shotgun.
JR: No.

HC: But it's a condenser, which is potential battery and mike-outage problems . . .
JR: We use AC power supplies. There's so many things you can do under the right circumstances. On the TONIGHT show, the boom doesn't move much.

HC: I assume they use the same boom for the monologue as for the guest shots.
JR: Right, and he's pulled back 6' and locked off – that's where he is for the rest of the show. In a dramatic show, that boom may be moving all over the stage. In a musical variety show, he's pulled back, he's in, he's out, he's really moving around. We still use condensers.

A battery problem and an AC problem is not as big a problem as RF. If you tape the power plug in the socket, are careful with the power supply, and treat it for what it is, your chances for failure on the air are very small. But the reason I don't use condensers on a show like THE MIDNIGHT SPECIAL is; we don't have *phantom* power supplies. Now those power supplies are down near the stage where we have hundreds of people milling around or passing nearby and it's easy to lose a power supply.

HC: How about pre-production and rehearsals?
JR: Pre-production . . . I think I would like to see more meetings with some of the engineers, the kind of meeting that will allow us to know what the total "package" is going to look like. I think the mixer should know what the "look" of the show is to be . . .

HC: The cosmetics?
JR: Right, and maybe he won't have so many problems in his head. He can go in and say, "I know, because of the look of the show, they're going to want to use booms." He won't spend a lot of time getting himself psyched-up to hand mikes, and then they say, "No, it's a boom." He can know *before* and think, "I know they'll want to use booms because that's

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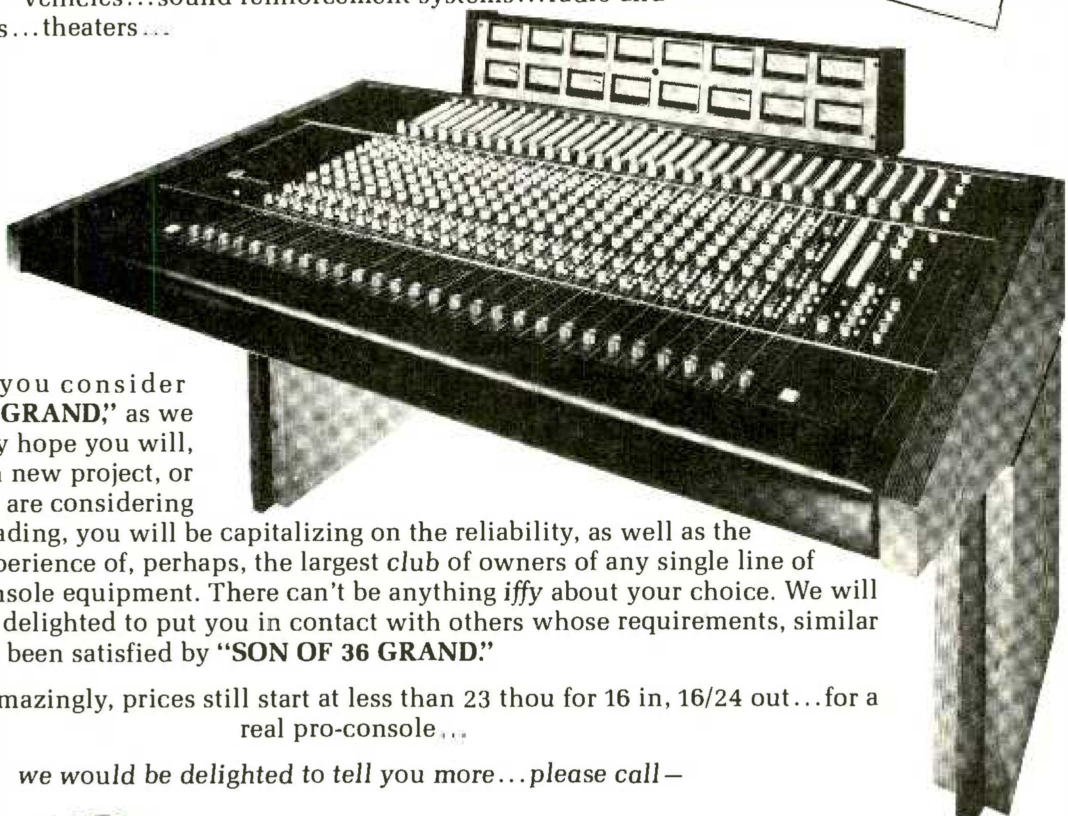

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Joe Ralston at NBC-Burbank

a fantastic set and they'll want a shot of that whole thing."

I know some people treat us as mechanics and we *are* mechanics and technicians but that doesn't mean we don't have a "feeling" of what is of value. I don't mean we should be consulted or should infringe on the direction or production but I would like to have "feeling" of what *they're* after *before* the day of the shoot. But to come in at the last minute after they've got everything set up their way; even if you *wanted* to change something and they agreed to change it, it's too late because the set is built this way and everyone's rehearsed it this way. I just think I would like to have an idea a little sooner just for my own good. If I knew a week ahead, I would plan for it and figure out how I would go about achieving it.

HC: What sort of relationship exists between you, the camera man, and the boom man in the sense that you want the boom man to go in to pick up better sound, yet the director may not want him in too close otherwise the camera is going to see him.

JR: Right. We compromise. We lose an awful lot.

HC: Audio loses?

JR: Yes, because it's for the same reason they don't want to see a hand mike.

HC: Who has jurisdiction over the boom man? The director or Technical Director (T.D.)?

JR: The T.D. has jurisdiction. He is in charge of all engineers of that show – audio, video, and lighting but he allows them all to sustain themselves – to take care of themselves. If there's a big problem, they'll try to hash it out. Most Technical Directors allow the audio man to do his job, etc.

When it comes to something like you're talking about: if I've got a problem, if we're not getting enough level, the T.D. will discuss it with the director and tell him the problem and the director may say, "Tighten the shot – get in there *or* at such-and-such a point I'll cut into a tight shot and you can move in." Now if the director says, "*I* think it's OK", *now* it's a question of what *I* think is right and good and what *he* thinks is right and good. But *he's* in charge.

I may hate it, dislike it, froth at the mouth, but there's nothing I can do. I can argue maybe and ask for it to be a little tighter and maybe I can win, but if he wants it, I'm stuck, that's the way it is.

HC: Do you feel that most "professional" directors are cognizant or sympathetic of audio problems?

JR: Some are and some aren't . . . some are interested in one thing. I have another feeling on that. The director is in charge of the entire show. But for some reason when that director credit rolls by, *I* feel everybody thinks, "What a great job he did with his *pictures.*" They don't think, "What a great job the director did with the *sound.*" They wait until the audio credit goes by and if the *sound* was bad, "Oh, so-and-so did it." If it was good, "Oh, teriffic." Maybe the director could have been the guy who actually came in the booth during a big musical number and said, "More saxes, more guitars, or whatever." He could be very qualified musically and played a very important part in the sound balance. But still when his name goes by, I don't think many people relate *sound* to the *director*. Consequently, when a director does a show, he doesn't care that much in most cases. Sure his name is on the crawl (credits), but it's not associated with audio as it is with camera shots. In most cases they'll go with that good camera shot if it comes to a choice.

It was always good to work with the production company on THE MIDNIGHT SPECIAL. They were concerned about audio and they always asked how we could improve it. They always put audio on top.

HC: What about communications between mixers, T.D.'s, and the set? The PL's (private lines) or cans?

JR: A pain in the neck! Anything you have to wear. I sympathize with dramatic shows and situation comedies; the mixer is in constant communication with his boom man. It's changed now, but for years, you had to wear that PL on your head to talk to him or to P.A. First of all, it covers one ear and it just hangs there and if there's talking back and forth between other guys, it can be distracting.

They've come a long way from that: now you can talk into a little mike and it will be communicated to the boom man's PL.

HC: Could you contrast doing game shows or a sitcom vs. a music show.

JR: I'll in no way put down a game show or sitcoms. I have done game shows that have been more complicated than some of the big specials. When I did LAUGH-IN, on Mondays, Tuesdays, and Wednesdays, there was so much post-production involved and a lot of pre-recording involved. As big a show as it was, it was basically a two-mike show. Most of the sound effects and music were laid in and laid in.

Thursday and Friday were for DEAN MARTIN. We had a live orchestra and because we did little rehearsing, never prerecorded, and did everything live, you're mixing a lot of mikes and there's a little more pressure. Whether it's right or wrong, someone from the production crew or talent may say, "That's it. I've had it. I'm happy, if you didn't get your thing done, I'm sorry." Or, you could do a thing like HOLLYWOOD SQUARES: a game show with two contestants, nine stars, an MC, live announcer, music, sound effects, audience reaction, everything going at once, *and* a problem of P.A.; Stars hearing contestants, contestants hearing stars and not themselves, mix-minus feeds, and it's just a game show. You crank out five a day, but it's very complicated.

I recently finished the RICHARD PRYOR SPECIAL and the music for all of that was done outside of NBC. With the exception of the Pips, there was nothing really involved — play-ons, play-offs** and a little background music. That's a comedy show.

Situation comedies can be difficult: two booms and off-stage mikes, etc., but you've got another problem with shadows, getting around the lighting, presence, perspective — all those things. They're totally different.

Doing a music show like THE MIDNIGHT SPECIAL, I don't worry about boom shadows. It's more like recording — but I don't want to say it's like recording. I've probably talked to and discussed audio with more managers, agents, recording engineers, road PA men, truck drivers — everyone who goes on the road with a band. I've had people tell me how to mix a band from the manager to the producer to the guy who drives the truck. And every one of them says, "I've mixed so-and-so's album". I deal with all of these people and they tell me about recording and how it is and how it should be, and to this day, it's still not the same.

There's things we have to do that are totally different. A lot of things happen with television that probably don't happen in the recording studio, like an audience. But we just don't seem to get the discipline for some reason.

HC: Of the performer?

JR: Of the performer, of the musicians backing him. When they go to television, it's a farce right off the bat. This is the attitude I feel they have. "The hell with it. Let's do it and get out of here." They go into a recording studio and say, *"Hey, we're going to cut an album."* Everybody's really into that. But television is a different thing. We're fighting that battle when they come in along with the battle of television sound.

HC: Do you surprise a lot of them by the time they leave – for the better?

JR: We have. We've had some very good things come out of the show. I think I've made some good friends who've come in with bad attitudes for television. I've probably sent a lot of people away saying, "We knew it", and I've probably sent an awful lot away saying, "This wasn't so bad".

A lot of people think we do these big mix-downs and it's a whole big thing. Well, THE MIDNIGHT SPECIAL is not done this way. We take a lot of guff in putting it together.

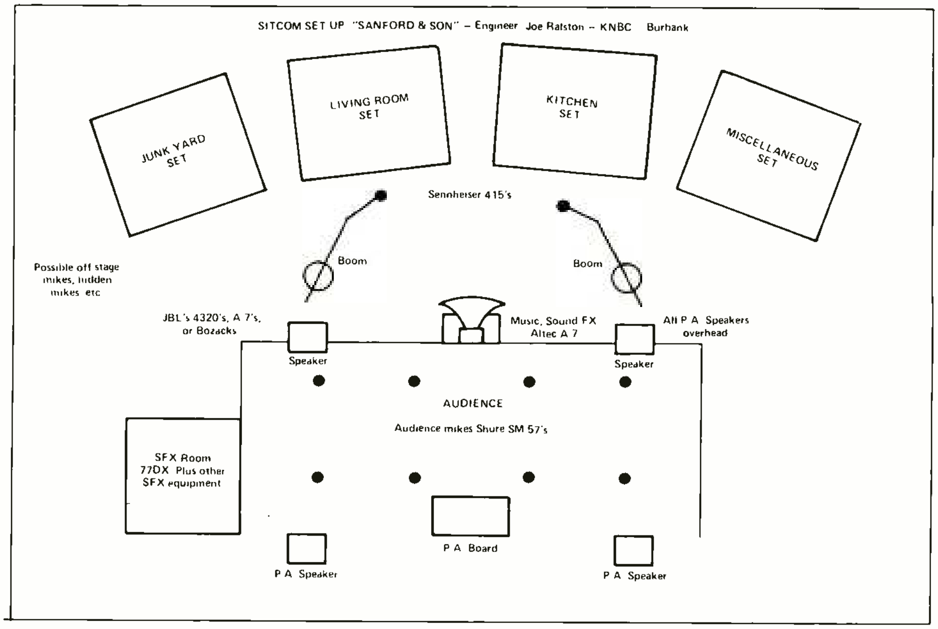
People take people out of the recording industry with big hit records and bring them into television under a time limit and pressures to try to do something that will turn out half-way decent and it becomes a grind after awhile. But I think in television today, I would rather do that kind of show – because of quality.

HC: Do you have people coming up to the booth telling you how to do music?

JR: They do. We've had them limited. I like listening to their ideas, but after awhile you're dealing with a lot of different personalities. I don't like pushy people and with the union here you can't have people pushing knobs. I'd *love* to let them mix the group because they probably know them better. But I've had a lot of people go away mad – and a lot of people understand.

HC: What about the boards?

JR: At NBC, the boards are built here.



When they started, they *did* talk to engineers to ask them what they wanted and took that all into consideration: to record big groups and have all the inputs. We could do just about everything that television needed. They felt, "Let's go way out" and "What if?" They put things in the board we thought we'd never use and they *didn't* put things in the board that we really needed. Who thought that we needed solos? We all need solo, but we can't do it.

HC: What features would you like to see in the board besides solos?

JR: Equalization in the mid-range. We don't have the extensive equalization that the recording industry does – to push this here and push that there and "stack" it or layer it. Extensive equalization, if you have the time, really tailors something and makes it very nice, when going in, you thought it was very bad.

We now have EMT's and I'd also like to see digital delay. We get them but I have to rent them. I also rent Gain-Brains and special microphones.

HC: When you EQ, do you EQ something taking TV sound into consideration, or EQ it as if you were making a record?

JR: That's probably my biggest argument with most people that come in. We have JBL 4320's we're listening on, but what they're hearing on them is not going to happen at home, so I like to EQ with a TV speaker – any tiny speaker I can get a hold of. I have an Auratone that I use . . .

HC: . . . and A-B back and forth?

JR: That's exactly what I do.

HC: Do you find yourself boosting the bottom end for TV?

JR: That's one of our problems with EQ facilities on the board. On the bottom end we're dealing with 100 Hz – below that, forget it. On the top end, there's 3k, 5, and 10k knobs. We have nothing other than that to work with.

HC: I thought you'd maybe bring in an outboard for THE MIDNIGHT SPECIAL.

JR: No, we don't. That would really be something. We have graphics, but there's so many groups and problems that it would take a week to do the show instead of a day or two.

HC: We were talking about doing sitcoms and game shows vs. a music show and its wider frequency response.

JR: Right. There's an opportunity for wider frequency response and sound but there's a problem in what you do with all of these sounds.

If it's a music show, I try to "stack"

them in their order of importance to see if they can fill the whole band. If there's an instrument, you've got a picture to contend with, and if you don't hear it, you've got a big problem. The arrangement may be written without that instrument as a solo in mind, but the director has the correct light and wants to feature that sax or whatever, and fill the screen with the sax player. The people at home may say, "I don't hear the sax". Well, the tune wasn't written for the sax! I may have someone standing over me saying, "Don't bring up the sax", but yet . . .

HC: . . . the director may want to feature the sax.

JR: Right, so we've got to go back and talk it over. Maybe the sax isn't playing that big a part. Still the director wants it and I have to go along with it a little. It's got to go with the picture.

If I didn't feel that way, then I really shouldn't be in television, I should be in a recording studio. The picture has to go with the sound and the sound has to go with the picture.

HC: So against your better judgement you may feature the sax.

JR: Exactly. I would make him heard, even for a little bit and then bring him back down. Many times I've been watching the TV – stuff that *I've* done, and I feel that should be louder – it should be heard a lot more.

HC: What about multi-track machines?

JR: I think they came into being – the four tracks and up – around the time of the ANDY WILLIAMS SHOW ('68). Now I use a 16 every week for THE MIDNIGHT SPECIAL. When I did DEAN MARTIN and LAUGH-IN, I used 4-track, and BOBBY DARIN was 8-track in '71 or '72.

We wanted to do simulcasts for THE MIDNIGHT SPECIAL. That show allowed us to move into 4-track, then we got into 8-track to alter things.

The production company might promise groups a lot of things to get them to appear on the show. We got a few in that insisted we get 16-track or be able to do certain things. The first 16-track we used, I had to rent. It was an advancement we never had before – in a mono medium.

HC: What sort of tracks would you use in production and what sort of tracks would you leave open for post-production?

JR: We don't do that kind of thing here. What we do is use the multi-track for pre-records or stage production. We try to mix the composite using the multi-track in the studio while it's being done and put the composite on video tape. Then once it gets to post-production, we use an 8-track in post-production, and put the original video tape tracks down and the time-code on another track. When we use the original videotape track we put it through whatever equalization it needs.

HC: This is audio on videotape?

JR: Audio on 2" 16-track using only 8 tracks – the tracks are staggered to prevent bleed-through (crosstalk). The original audio track goes through EQ then goes to another track and is known as an "adjusted videotape track". Now we lay sound effects, music, announcers, looping, audience, all on separate tracks and then remix the whole thing to a composite.

We stagger because of the leakage of the tracks and the EECO time-code, which is a variable frequency. From that we can pull out our sync of 59.9 Hz for resolving.

HC: Is there such a thing as video alignment tones on the head of a reel?

JR: We put multi-tone on everything. It's frequencies of 50, 100, 400, 1k, 3k, 5k, 10k, 15k. It's in every studio and comes up on our trunk line and we just set level to that.

for additional information circle number 20

HC: What about alignment of audio equipment?

JR: In recording studios it's the thing, but generally television doesn't do it every day – there's not enough time. If we feel something going south, we get out the alignment tape and look at it.

HC: Can you use the multi-tone to check your equipment?

JR: Right, that's generally the way – just look at the meters on the machine and check input vs. output and if it looks good, we're in the ball park.

Television sound doesn't have the quality control that the recording industry has.

HC: Is that because of the human shortcomings or lack of people pressing QC?

JR: It's sort of, "So what, it's good enough for local". We run alignment checks when we notice something is out. For MIDNIGHT SPECIAL we run an alignment check. It's not a practice because there's not enough time. So I think our quality has gone down the toilet because there's not enough time and budget.

HC: Do you find that you get good tech back-up or support?

JR: Generally there's never a problem. If I need a piece of equipment, I can get it or rent it. Most of the supervisors have all been through TV problems and realize yours.

The changes in TV audio have been very fast – from single miking to multiple miking, etc., and the tech supervisors that exist today were most likely tech supervisors at that time and consequently may not have the experience of modern-day recording techniques. *But* they have the confidence in the mixers and are sympathetic and understanding to the extent that they will let you do what you want.

HC: Any comments on "curving" the stage?

JR: We do that on MIDNIGHT SPECIAL and we use 1/3 octave filters with Soni-Pulse in our feedback systems but we have another problem in that the monitors are placed in so many different areas, depending on the groups, and unless you Soni-Pulse each time, and lock down the speakers, it doesn't make much sense. And we don't have the time to break down, Soni-Pulse, draw a curve and re-EQ it. If it's a *critical* problem, then we make the time. But if we can get by, we don't spend the time because we don't have it. The important thing is to get the show on and get it going.

HC: The overriding picture I'm getting is the "time" factor.

JR: Right. There's too many people waiting on you so the first thing you do is get it going, *then* if you have the luxury of improving it, you can fool around. After working with a show for awhile, you can set up quickly, and then take care of those things which are of a lesser priority.

Some people get a regular show, they do a set-up and stick with it. I get pretty stagnant if I do that. On MIDNIGHT SPECIAL, I'll experiment. Maybe one week I'll try a different drum set-up or experiment in a game show, using a certain amplifier set-up to feed something and then change it the next week.

HC: What about the signal processing and EQ you go through for the various shows?

JR: We have tape noise problems, a lot of that hiss. By the time the tape gets out on the air, it has gone through a lot of generations. We have 5 cameras. We have a lot of studio noise, not only with the audience, but with 5 cameras driving their cables across floors. We have 15 or 20 stage technicians, doors opening and closing, air conditioning. So now we're into high-pass and low-pass to take a lot of the rumble out. For example, the fan in the camera could drive you crazy.

HC: How close are the cameras to the action?

JR: In dramatic shows – about 5 feet. When you have a quiet stage with tile floors, you can hear a camera cable 50' away. And if the production stretches from one end of the stage to the other and you have live transistions, you always have a build-up of noise.

We have a big problem with cables on vinyl floors. This lends to high frequency build-up and consequent pops and cracks from just walking on tile floors. We try to wrap the cables in cloth but you still get the noise.

HC: Any other special effects for TV production besides digital delay and phasing? Voice filtering?

JR: Oh, yeah. We use voice filtering for television effects, intercoms, or telephones. We use reverb to simulate P.A. in a night club, or a prison, or a court room scene. We use a lot of sound effects.

On LAUGH-IN, we used *tons* of sound effects. A lot of them were done live, but most of them were cut in. NBC just finished OUR TOWN (Hal Holbrook), which was a *big* undertaking in sound effects.

HC: Does a lot of compression take place?

JR: This is something we're concerned with. Here, our main concern is the transmitter for the local stuff. When you do a show, the masters can be sent and aired in New York. There are so many stages before the transmitter that you can have problems with – videotape machines, alignment, is there a good master to work with?

Also, there are a few transmitter engineers that really run scared because of the FCC and set their compressors really tight. There may be a few spikes with audiences or effects and if the guy doesn't want to take a chance, you'll have the pumping. Maybe a tape goes through a lot of electronic editing and generations, and the levels could change through each transfer. The equalization could change and suddenly you may have a lot of top-end which can play hell with a compressor.

The television speakers like to push 5 kHz and make it bright. We mix with JBL 4320's and in post-production we have contour controls. We've used Soni-Pulse in the control room at times, but you can spend a lot of money and it's still coming out of that little speaker. It's just not justified yet to the people that make the show run.

I listen to THE MIDNIGHT SPECIAL while recording and then listen to it in post-production and it's two different sounds, maybe four generations down, but still the same speaker within two different rooms. We use the television speaker and I find I can shove a band track right up behind a singer. Many people will come up to me and listen and say, "I can't hear the words". So I started putting it on a little speaker and suddenly you hear the words.

When I was working on BOBBY DARIN, the band sounded like it was so far in back of him on the TV speaker, like they were across the street. When I would listen to it in the studio and then on the television set, there was a great difference. Then I started pushing the band higher, but so that you could still hear the words. *Then* when I listened to it, I could hear the presence.

As far as filtering, I roll off anything below 60 or 100 Hz because all it does is move the meter. With dialogue, I roll off at 5 kHz because of all the electronic and other noises around the stage. Music, of course, has the widest range. I do use notch-filters as outboards for fans, etc., and Kepexes a lot on sitcoms and dramatic shows. I also use what we call "snappers"* – just putting a signal out of phase to cancel background noise but not enough to hear the out of phase sound.

HC: Do you have anything to say about audio in post-production stages?

JR: Yes. I think that each mixer who did the show should be the mixer in post-production but sometimes it doesn't happen that way.

---

* Controlled through level. Two limiters, with the phase reversed between the two, are set up. One limiter is set approxmiately ten times harder than the other. Two separate subs are sent to the same channel and the second fader introduced until the cancellation is heard – then backed off. This helps in cancellation of studio noise.

HC: How about if it's already been committed to videotape; why would you need to do additional mixing?

JR: That's just it – *you* may have recorded it and know . . .

HC: . . . to correct that shy bass line

JR: and suddenly you hear it back. They're not going to do it again – the set has been torn down – but you say, "Gee, if we did it once more I'd boost a little EQ". *You're* there, *you* know, and *you'll* do it. You've got that pride and you'll make it the best you can.

I've read a lot of books, attended a lot of seminars, listened to a lot of people talk, and listened to a lot of theory, but to me, I just won't accept *it* as the gospel. I'll try it and experiment with it, then make my own decision. I'll try almost anything. I've tried some pretty crazy things to get the results I'm looking for.

You know these brochures that say "Such-and-such is good for such-and-such and so-and-so mike has a curve for so-and-so and you've got to . . ." Well, I don't follow that. I'll listen . . . but I won't be influenced by what the books say or what the manufacturers say. *I've* got to try it. Maybe it *won't* work, but maybe for television or what *I* want, it'll do the trick.

□□□

With all these conflicting reports about our 65K Programmer, who can you believe?

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ALLISON RESEARCH HAS ASKED FOR AN ENDORSEMENT FROM ME CONCERNING THEIR 65K AUTOMATION PROGRAMMER SYSTEM. THIS IS SOMETHING I'VE BEEN RELUCTANT TO DO IN THE PAST FOR OTHER MANUFACTURERS AS I FEEL ONE MUST TELL THE BAD WITH THE GOOD AND NOBODY LIKES TO PRINT THE BAD IN THEIR ADS. THIS TIME, HOWEVER, THE BAD IS GOOD. BUT FIRST, THE GOOD.

RELIABILITY. AFTER SELLING AND INSTALLING OVER A DOZEN ALLISON PROGRAMMERS IN CONJUNCTION WITH HARRISON CONSOLES, I CAN GRATEFULLY REPORT THAT THE FAILURE RATE HAS BEEN ABSOLUTELY ZERO. FOR AN OLD, ESTABLISHED PRODUCT THAT'S VERY GOOD. BUT FOR A PRODUCT WHICH IS BOTH RELATIVELY NEW AND EXTREMELY SOPHISTICATED, THAT'S UNHEARD OF IN OUR INDUSTRY.

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NOW THE BAD.

THE MAINTENANCE DEPARTMENT WILL HATE IT. ABSOLUTELY HATE IT. AFTER ALL, THEIR JOBS DEPEND UPON EQUIPMENT BREAKING DOWN (PREFERABLY ON OVERTIME) AND THE ALLISON AUTOMATION PROGRAMMER SIMPLY REFUSES TO COOPERATE. VERY BAD. VERY UN-AMERICAN. IF ALL MANUFACTURERS START TO BUILD THEIR EQUIPMENT THIS WAY, OUR UNEMPLOYMENT RATE WILL RISE AGAIN, OUR TAXES WILL GO UP AND MAINTENANCE PEOPLE WILL HAVE TO FIND AN HONEST WAY OF MAKING A LIVING. PAUL BUFF, WHAT ARE YOU DOING? YOU ARE SETTING A VERY DANGEROUS TREND.

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Allison Research
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Nashville, Tenn. 37204

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Sincerely,

Norman Dlugatch
Technical Director

ND:lk

Harrison

Paul Buff, President
Allison Research Inc.
2817 Erica Place
Nashville, Tennessee 37204

Dear Paul,

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Kindest Regards,

Dave Harrison
President

HARRISON SYSTEMS, INC.

GUGGENHEIM PRODUCTIONS, INC.
3121 SOUTH STREET, N.W. WASHINGTON, D.C. 20007
202-FEDERAL 7-6900

Mr. Paul Buff
Allison Research, Inc.
2817 Erica Place
Nashville, Tennessee 37204

Dear Paul:

It was good to see you in Springfield a couple of weeks ago. Since I have not received any phone calls from Henry, I assume all went well at the start of their season.

In response to Norman Baker's request, let me offer the following:

"When we set out to design the Springfield, Illinois, Sound & Light Program, we searched desperately for programming equipment which had enough capacity and flexibility to do the job: a 45 minute program, six audio channels into any combination of ten playback channels, 150 lighting circuits operated by 35 dimmers in 100 load switching relays, etc. Then, we wanted the ability to program the show on location in real time using dimmers and faders. Most importantly, we wanted equipment reliable enough to let it operate without an electronics wizard hovering behind the rack.

"Allison Research's 65k-2 Programmer came along just at the right time -- and at a price we could afford. The equipment did everything that had been promised - and continues to do it. The program is now operating in its second season and we have yet to lose a showing due to equipment malfunction (once they get the weather control wired into the programmer, the record will read 100%).

"There was a bonus we received when buying the Allison equipment: Paul Buff and his staff brought a professionalism and commitment to our project which I call first rate and which made everybody else pull a little bit harder too.

"I shall welcome the opportunity to do business with Paul again when it arises."

I hope the above is helpful.

Best regards to all,

Werner
Werner Schumann

WS/esg

for additional information circle number 23

sigma sound studios

Mr. Paul Buff
Allison Research
2817 Erica Place
Nashville, Tenn. 37204

Dear Paul:

Sigma Sound Studios was hungry for automation long before you sold us our two Memories Little Helpers back in '75.

The MLH system eased our appetite. We became dependent on Memories to the point that "glitches" and other tape related problems were academic compared to the creative freedom gained with the system. You know that both our Memories have been in continuous, 12 hour/day, operation since they were uncrated.

Now with our three (and fourth on order) 65K programmers we sit back – quite content! We have retrofit two of the programmers into our MLH systems, the third has been incorporated into the first of our custom MCI consoles fitted with your digital faders which we had built for our New York studio.

Paul, the point is this: we have eighteen months of cumulative time logged on the 65K, and it has performed in one word – flawlessly. Not one "glitch", "stutter" or "drop out" – nothing but reliable operation. Crosstalk and data leakage are non-existent. In fact, the governing factor as to our data level seem to be keeping it visible on our deck meters (–20db). The propagation delay of the 65K programmer is unappreciable, our current "record" stands at 33 passes made with our memory plus system – mutes written on the first passes still came up on tempo!

With the 65K programmer, automated mixing at Sigma Sound Studios is a beautiful, day to day, reality.

Thanks Paul !!!

Sincerely,
SIGMA SOUND STUDIOS
Wayne E. Wilfong
Wayne E. Wilfong/Director of Engineering

DIFFERENT FUR MUSIC

Norman Baker
Allison Research
P. O. 40288
Nashville, Tenn.
37204

Dear Norman,

Just a note to thank you, Paul, and the rest of the crew for the new programmer. It works! Not only that--which as programmers go may itself be surprising to some--it always works. We do all the things you're not supposed to--punch in and out on the data track, put the data on edge tracks, run the tape ragged, and so on, and so far it doesn't know the difference.

Happy to have you use our name by the way; we'll be in good company, I'm sure.

Sincerely,
Pat —.
Pat Gleeson
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Paul Buff
Allison Research, Inc.
2817 Erica Place
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NORM BAKER
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2817 ERICA PLACE, P.O. BOX 40288
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a comparison of the techniques and duties of the people involved in the process of audio

Post-Production Sweetening

for VIDEOTAPE and FILM

by
Paul Sharp

Post-production film sound has been around for awhile. By comparison, audio sweetening in videotape, at least as it is now being practiced, is rather new. Besides the networks, the houses in Hollywood that are doing much sweetening can be numbered on the fingers of one hand. As the use of videotape grows, however, the field of post-production audio sweetening will also grow, and more of it will take place as possibilities enlarge and methods improve. This article is intended to consider some goings-on in sweetening in the Los Angeles area, compared to the film post-production sound process. Film and television are types of businesses difficult to compare, since the nature of any single production may call for very different methods from that of another, in addition to the general differences that exist in the nature of the program material that is common to each medium. Since sweetening is relatively new, we shall be comparing film from where it has been for some time, to tape where it is just (again, relatively speaking) arriving.

The following block diagrams should be helpful in examining the two processes. Let us assume we are looking at a filmed-for-television show (Figure 1) and a videotaped show (Figure 3). So that the whole process can be shown, the block diagrams include the steps followed for a musical type of production, even though a musical variety production is usually taped, rather than filmed. The "produc-

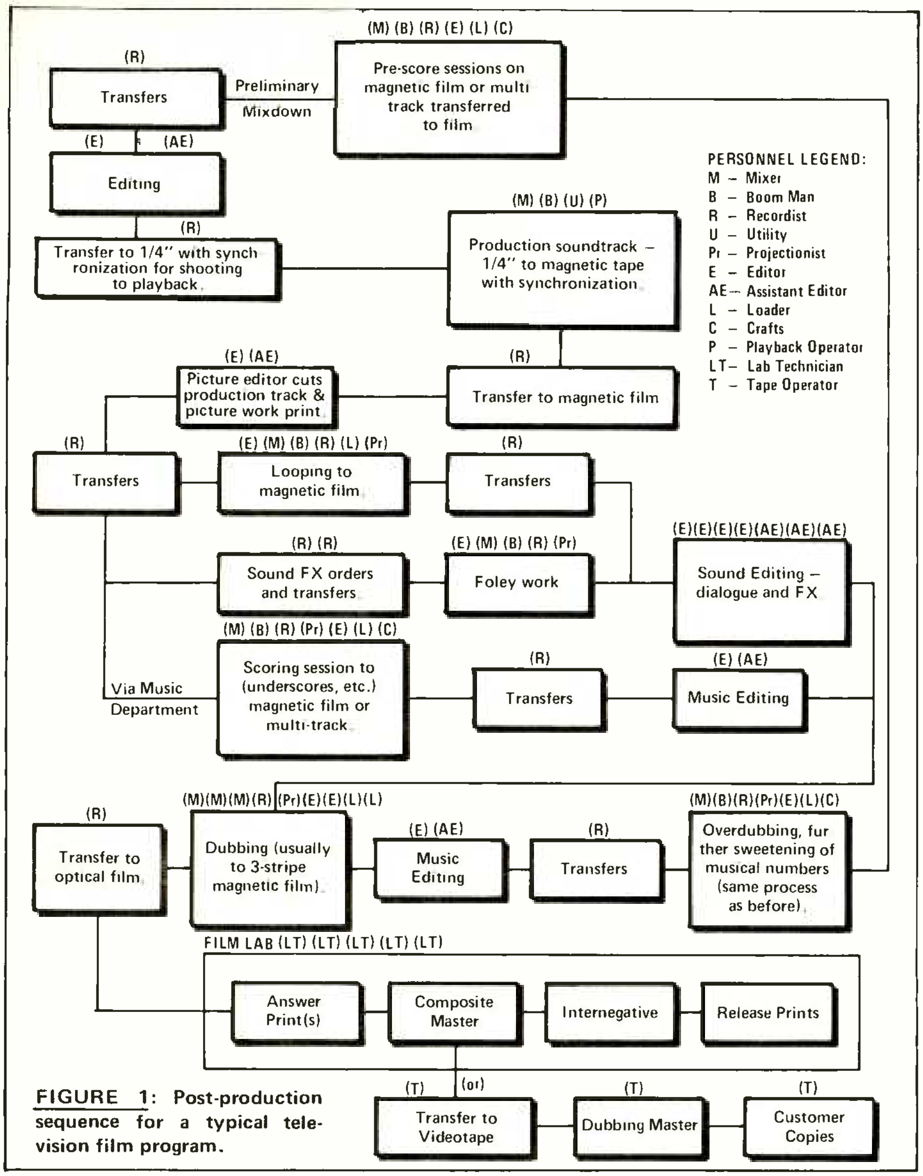

FIGURE 1: Post-production sequence for a typical television film program.

tion" steps are included because of their relation to the "post-production" operations.

The circles above each block represent the number of men typically involved in that step of the operation, with a key explaining the letters used. The diagrams and titles are intended to be representative, not "hard and fast". It should also be pointed out that some of the personnel shown in the block diagram might be the same person(s) in a later step of the diagram. Examples are the Music Editor and his assistant, transfer recordists, etc.

FILM PRODUCTION PRESCORE SESSIONS

If the session is being recorded on 35 mm magnetic film only, only one recordist may be used. However, the session may also be recorded on 16-track plus a ¼" or ½" back-up copy. Another recordist or even two more recordists will be used in such cases. A couple of studios use two mixers for standard scoring sessions. One might handle the recording, and another handles a "playback" of the recorded tracks, through a separate playback console. The author even knows of a studio that uses three mixers.

In film work, three tracks might be recorded on one piece of 35 mm magnetic film, and that will be used as a playback cue while another three channel recording is made, which is in turn played back along with the first while another is made, and so on. In this manner, film music was in the "multi-track" business long before the recording industry.

The boom-man, or microphone operator, handles the set-up and arrangements on the stage, under the direction of the scoring mixer. He will have some help, such as a "loader", or cable-man, and usually someone from another craft union who assists in non-sound functions (chairs, music stands, etc.).

The music editor(s) is always present for the session. The music editor may well be present for the dubbing session, too, when that stage is reached. This editor will cut the film (or rather a copy of it) into "playback" format. The labeling here is "preliminary Mixdown", since the mix usually ends up being redone after further sweetening.

TRANSFERS

Transfers will be showing up a lot during our discussion of film post-production sound. Original recordings are rarely cut themselves. A copy is made, and the copy, called a "transfer" is what is actually cut or otherwise worked with. In this first example the preliminary mixdown is copied to another piece of 35 mm magnetic film, and this copy is used by the editor to create a "playback" version.

EDITING

The music editor (and his assistant, not to be forgotten) cuts a "playback" version of the musical numbers. He usually cuts a couple of bars of "clicks" from a "click track" onto the head of the number. These are nothing more than click sounds in tempo with the music. These clicks, incidentally, are usually used in the original sessions as a cue track (fed over earphones — "cans" for you English types) to assure that a strict tempo, hence timing, is followed. For the playback copies, the editor cuts some clicks on the head of each number, and often between sections of the music, so that a convenient tempo reference will be heard upon playback before the music actually begins, since actors will be synchronizing their actions to the music.

TRANSFERS

The editor's playback version on 35 mm magnetic film is copied onto ¼" tape, with a "sync" pulse, usually 60 cycles (no, the author isn't that old — he just has an aversion to naming something after a man when a good label already exists!), and this copy (plus a spare that is usually made) is used for the production filming to "playback".

PRODUCTION SOUNDTRACK

When music numbers are being filmed (this is assuming that they are not being recorded live, which is rare for film), the pre-recorded music that now exists on ¼" tape is played back from a ¼" machine that will read the sync pulse from the tape, feed that sync pulse to a "resolver" which compares it to a known reference (the AC line reference or a 60 cycle — strike another blow for the non-scientific ego — crystal oscillator, etc.), and this resolver then feeds a motor control voltage to the playback machine, varying its speed so that the "playback" is at a precise and constant speed. The reason for all this is so that when the film is developed and the original music track (sweetened, or whatever) is played back with the film, everything will still be synchronized. We are assuming that the camera is also running at a constant speed, and although there are other ways of accomplishing the job, and different methods of synching the playback and the camera, I


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won't go into all that here. Which reminds me, I neglected to point out that when the music is originally recorded onto magnetic film, that film has sprocket holes in it, which "keep it in sync". (The playback is also recorded onto the production soundtrack recorder, which is out there working along with the playback machine, and this recorder also puts down a sync pulse signal onto its tape, and this tape is transferred, and used as a reference, for viewing "dailies", etc. Since all that doesn't really come under the heading of "post-production", we won't go into it further.

One reason for going into this sync pulse business is because a surprising number of people don't understand it — even some people who should, particularly in video — but more about that later.

TRANSFERS

(Here we go again.) As usual, transfers are made of the production soundtrack, which the editor cuts to conform to the picture cutting.

PICTURE EDITOR

The picture editor cuts the production soundtrack and the picture "work print" (a copy of the picture) to suit himself, the director, the producer, or the producer's wife, or whoever, and on we go to the next stage.

TRANSFERS

(I told you so!) Often, copies of the picture and editor's production track are made for other editorial departments (sound editorial, music editorial, etc.) — soundtrack copies for them to work on and use as references.

LOOPING

"Looping", in film terminology, is replacing bad dialogue with new good dialogue. The dialogue may be "bad" for a number of reasons: bad background noise, bad performance, bad voice, etc. Whatever the reason, it is replaced as necessary. In early days (some studios still do it this way) an actual loop of magnetic film was made by the editors, with a matching picture loop. As the loop went round and round, a sentence (or portion thereof) at a time was recorded, over and over, until the timing and performance were just right. Newer, ADR (Automated Dialogue Replacement) systems exist now, but the old system still works quite well as far as the end result is concerned. If an ADR system is used, you can eliminate the "L" above the "Looping" block for the Loader. Sometimes looping is done to sound only, in which case you can scratch out the projectionist's "P".

SOUND FX ORDERS, FOLEY, AND TRANSFERS

Sound editors look at the picture, listen to the production track, and decide which FX need to be added or replaced. They then order the same from a library or order a copy of a good effect from the production track original tape, and begin their work. If the effect is not available otherwise, it will be recorded.

"Foley" will be done on a "foley" stage. This is a recording stage with pits of sand, water, brick, cement, etc., where experts will recreate footsteps, the rustle of clothes, bushes, and many other sounds as they watch the picture. These are all recorded, transferred, and cut into a dubbing unit, or reel, by the sound editors. Usually at least two recordists would be involved in all the various transfers.

SCORING SESSIONS (UNDERSCORES)

Here, the same technical things happen as in the prescoring sessions, only you can add a projectionist, who by now has picture reels to run while the scoring takes place. The music for underscoring could conceivably be done at the same time as the sweetening, for the music numbers, although I have listed it separately on the diagram.

(This step would probably be eliminated in most videotape shows, since scoring for them often comes from canned libraries, but if original musical scoring were to be done for a tape show, then this step would have to be included for tape as well.)

This scoring might be done to 35 mm magnetic film, to 16-track tape, or to a combination of these or some others.

TRANSFERS

Need I say more?

SOUND EDITING—DIALOGUE & FX

The FX cutting (editing) seems fairly straightforward, but a word or two about the dialogue work seems in order.

Film dialogue is the best there is, if the editors do their work well and if the proper looping is done (when looping is necessary). It can be terrible if editing is poor or if bad tracks are not replaced with looped lines. A production could typically include the following kinds of tracks, as illustrated in Figure 2, from a dialogue standpoint.

(1) The production track, with noises and other undesirable portions removed. The editor will actually physically wipe out the oxide coating, using a solvent solution on the film in a gradual sloping stroke, where necessary, to smooth out "bumpy" cuts.

(2) "Fill" of the background or room sounds. These are best cut onto a separate track, being cut in and out to the frame where the bad portions or bad noises on the production track were cut out. A separate reel may include a separate, continuous "fill", or "fills", to cover up the "holes" in the production track, although if the cut-in "fills" are correct, this won't be needed. Or, if the original background track was exceptionally clean, nothing may be needed to "fill", or only a slight "presence" track, or loop, at the most. A loop of "fill" may be made, which is run around and around in a circle on a machine in dubbing. Many stock sounds, such as camera noise, presence, outdoor presence, distant traffic, etc., are kept on

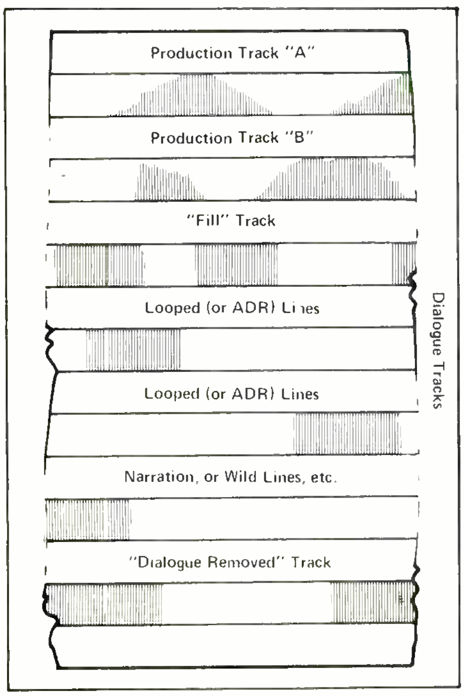

Figure 2: Possible channel composite for film sound and FX editing.

cartridges and film loops, and can be added whenever desired during dubbing.

(3) One or more looped tracks, or ADR (Automated Dialogue Replacement) tracks, where bad dialogue has been replaced. There may be also a narration track, or "voice over" track, or "wild lines" track (track containing lines that are not synched to picture).

(4) The removed portions of the production track are kept on another reel, and the mixer can bring them up on a fader at any time during the dubbing mix, should they be needed for comparison, or be deemed more desirable than the "fill" or replaced lines that were substituted.

The dialogue tracks are commonly split to an "A" and a "B" copy since different voices may need different treatment (EQ, filtering, etc.). This makes them easier to handle in dubbing.

MUSIC EDITING

Music editors cut underscores to fit the picture which, hopefully, the composer wrote and conducted to fit the picture, which, hopefully, hasn't been re-cut so much by this point that his underscores are no longer useable.

MUSIC OVERDUBBING, SWEETENING

The original prescore tracks may be sweetened considerably from the way they were done originally. The vocals may be entirely replaced. Or, if the director or stars (or whoever) doesn't like the original version, it may be totally re-done, using the old track as a guide (so that the new version will still fit the picture!). The final results are mixed down to magnetic film, which goes to:

TRANSFERS

Our old friend again.

MUSIC EDITING

The final music mixes are cut to conform to the picture, and the music units, or reels, are sent to dubbing.

DUBBING

The most common practice at film studios is to dub down to three tracks of a roll of 35 mm magnetic film. One track contains dialogue, another the music, and the third, FX.

A big film might contain 7 or 8 dialogue tracks, 2 to 4 music units (which would probably be three tracks each), and 10 to 20 FX tracks (or more). (If things get really busy, "pre-dubbing" will be done.)

Three mixers is common. One handles the dialogue, one the music, and another the FX. Big productions may use four mixers (two on FX). The picture editor is usually present, along with one or more other editors, and assitant editors.

Signal processing devices (equalizer, filters, compressors, noise gates, etc.) are used as needed to create the final sound wanted.

TRANSFER TO OPTICAL FILM

The final three track magnetic film is transferred to a roll of negative film on an "optical recorder". This machine exposes the film according to the modulations (created by the magnetic tracks) of a light valve . . . and so on, and I won't go any further into this, 'cause it gets technical, and 'cause I ain't up on it!

FILM LAB

At the lab, the optical soundtrack is printed along with the final-cut negative picture to a composite.

At this point, something created solely for TV (not likely for a film musical, but that's what we've used as our example) could possibly be transferred to videotape via a film chain, and copies dubbed for use by networks or independent stations.

Otherwise, "release prints" (film copies of the composite) will be made and distributed.

VIDEOTAPE PRODUCTION

PRESCORE SESSIONS

Videotape prescore sessions are typically done at a record recording studio, rather than on a film scoring stage, and in this setting, less manpower is used. The sync that should be laid down on one of the tracks is usually "vertical drive", 59.94 cycles, SMPTE time code, or both. This sync pulse signal is to serve the same use as the 60 cycle pulse used in film — when the beast is used for taping to playback, a resolver should be regulating the speed of the playback machine, and the process should be followed through right here from the beginning until the final music mixes are laid over to 2" videotape, so that sync will be good from start to finish. I should point out that sometimes the original prescore multi-track tape is used as a playback tape, or a multi-track mix from it is made (which should also contain the sync pulse) and this mix is used as the playback.

I have used the words "usually", "should", and so on, in the hopes that any readers who subsequently are involved in any of this type of work will assume that they should be doing this sync/resolving bit all along the way from step 1.

. . . continued overleaf

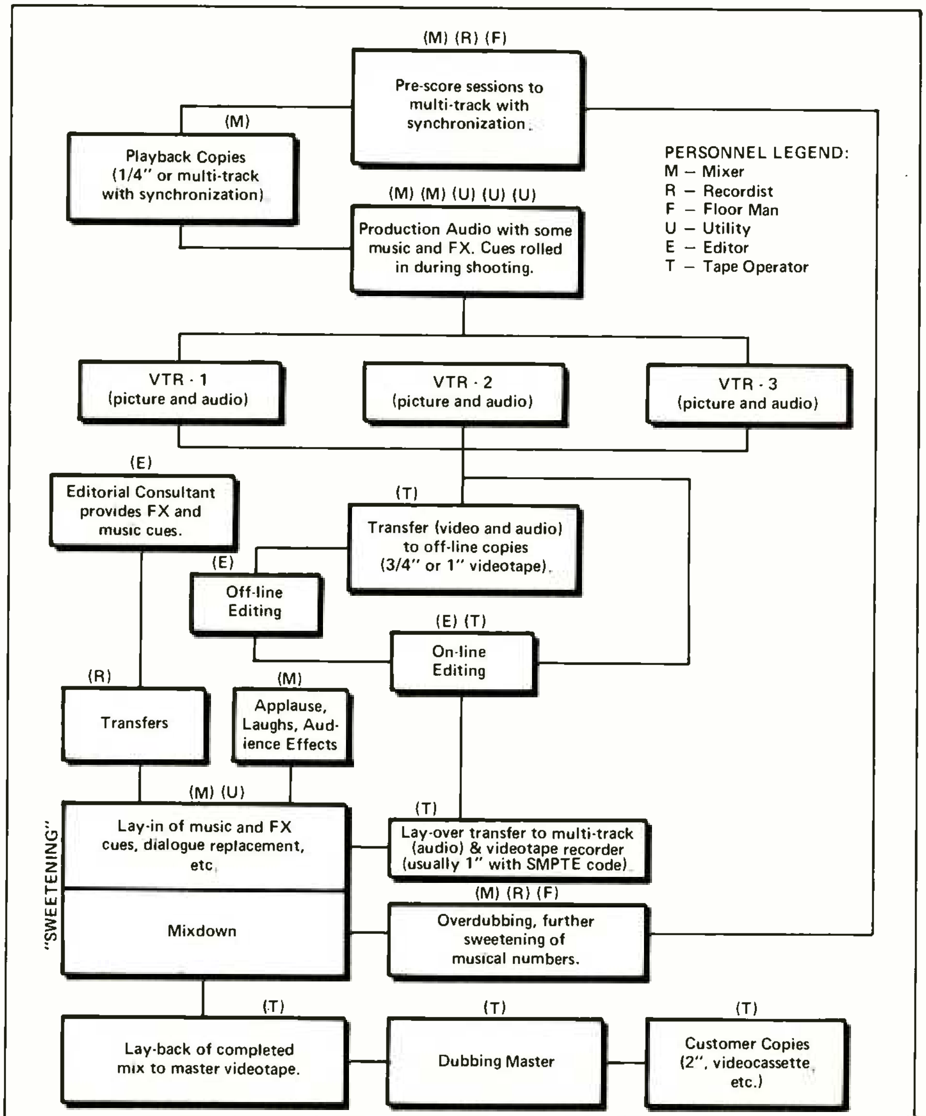

FIGURE 3: Post-production sequence for a typical television videotape program

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LEADER IN ADVANCED TECHNOLOGY

Unfortunately, many people don't understand the importance of this, and it would surprise the average reader if he knew how often post-production audio men (author included) have had to hand re-lay a music number — sometimes whole shows full of musical numbers — by the "slew" method with a VSO! Keeping sync, using the proper sync signal, operation of resolvers, and the associated procedures can be tricky, and if you are doing this type of work, you should get an expert from the beginning, or you might end up paying a lot more for an expert at the end and spending some agonizing hours trying to get things right!

PRODUCTION AUDIO

The idea of videotape has always been to get as much of the audio down correctly as is possible while shooting. For this reason, many tape shows never need any post-production "sweetening". When a tape show is shot, the opening music, closing music, announcer, FX cues, etc., may be all rolled in on cue by the mixer, or by an assistant.

But as post-production methods have improved, and as videotape shows have become more complex — especially musicals and musical variety shows — post-production "sweetening" has become more-and-more common.

There are still people around the country using film dubbing methods to sweeten their videotape shows. Others merely add what audio sweetening is necessary during "on-line" editing, using an audio machine and/or videotape recorders, live announcers on a mike, etc.

So when we start to talk about post-production audio sweetening for videotape, it's hard to discuss any particular method or procedure as if it were some sort of "standard" or common practice. As I mentioned at the outset, we shall consider "some goings-on" in the L.A. area, and that's about as close to being pinned-down as this article will get. But I shall try to cover, at least in general, the latest methods being practiced by the top houses in Hollywood.

VTR's

Besides the fact that the audio is recorded on the same piece of tape as the picture information, VTR's, or videotape recorders, by their operating nature, "resolve" during playback. They compare the signals (control track, etc.) on the tape with a known and very accurate reference souce such as the house sync generator, which is a very accurate, although costly little device. Hence, as long as the VTR's have a reliable reference, they will always play back previously recorded material at the correct speed. So the audio will not only be in sync with the picture, it can also be copied to another VTR and maintain synchronization.

No personnel are shown at this stage, since the operator(s) here is considered a part of the video, rather than audio, crew. Three video machines would be common for a musical show. No matter what picture information each would be recording, they would all normally be laying down the same audio track.

TRANSFER TO OFF-LINE COPIES

The 2" master tapes are dubbed straight across to smaller format working copies.

OFF-LINE EDITING

Video and audio editing are usually worked out with ¾" videocassettes, and a program of all edits is compiled which will be used to duplicate the process during on-line editing with the 2" master videotapes. The idea here is that the director and editor can sit down in a small facility with (relatively speaking) inexpensive machines and work out all their little editorial subtleties before putting the master 2" tapes up on the big machines in the expensive facility, and before working with the originals.

ON-LINE EDITING

After the off-line editing has been all worked out, the on-line process begins. If the off-line editing produced a good show, then the on-line process should be one of merely duplicating the off-line edits, using the original master videotapes. Of course, there is always someone who wants to do all of his editing in on-line right from the beginning. This is fine if the show doesn't need much changing from the way it was shot (and if the budget can afford it!).

LAY-OVER

The final edited master from "on-line" is copied back to a smaller videotape format, usually ½" helical scan, (although the latest EECO control system is programmed for the ¾" videocassette format as well). As the picture information is recorded on the small VTR, SMPTE time code, which was previously striped on the 2" master videotape, is also copied onto one of the audio channels of the VTR. At the same time, the SMPTE time code is also copied onto one channel of a 16-track audio recorder, while the edited *audio* track from the 2" videotape edited master is recorded onto the 16-track as well.

In other words, in one pass, the following are recorded:

1 – Video onto the small VTR.
2 – SMPTE time code onto the small VTR.
3 – SMPTE time code onto the multi-track audio machine.
4 – Audio onto the multi-track audio machine.

In actuality, a code generator might be "slaved" to the master 2" VTR code track, and its output fed to the smaller VTR and the audio machine, or the code track from the master might be put through a "restorer" and then fed to the other two machines, but suffice it to say that the code is striped onto the two machines from the 2" master.

OVERDUBBING

Musical overdubbing and sweetening are done with a video monitor (a beefed up TV set) in an overdub booth. The performers have the audio fed to them via earphones, and they add to or replace their former voices, being recorded on a track of the audio machine. Instrumental overdubbing might be done the same way. For multi-track musical numbers, the overdubbing could actually be done without any video reference, in normal recording studio style.

If video is desired, this is accomplished by having a synchronizer compare the SMPTE code on the VTR with the SMPTE on the audio machine and locking the two together while the overdubbing is done.

SWEETENING

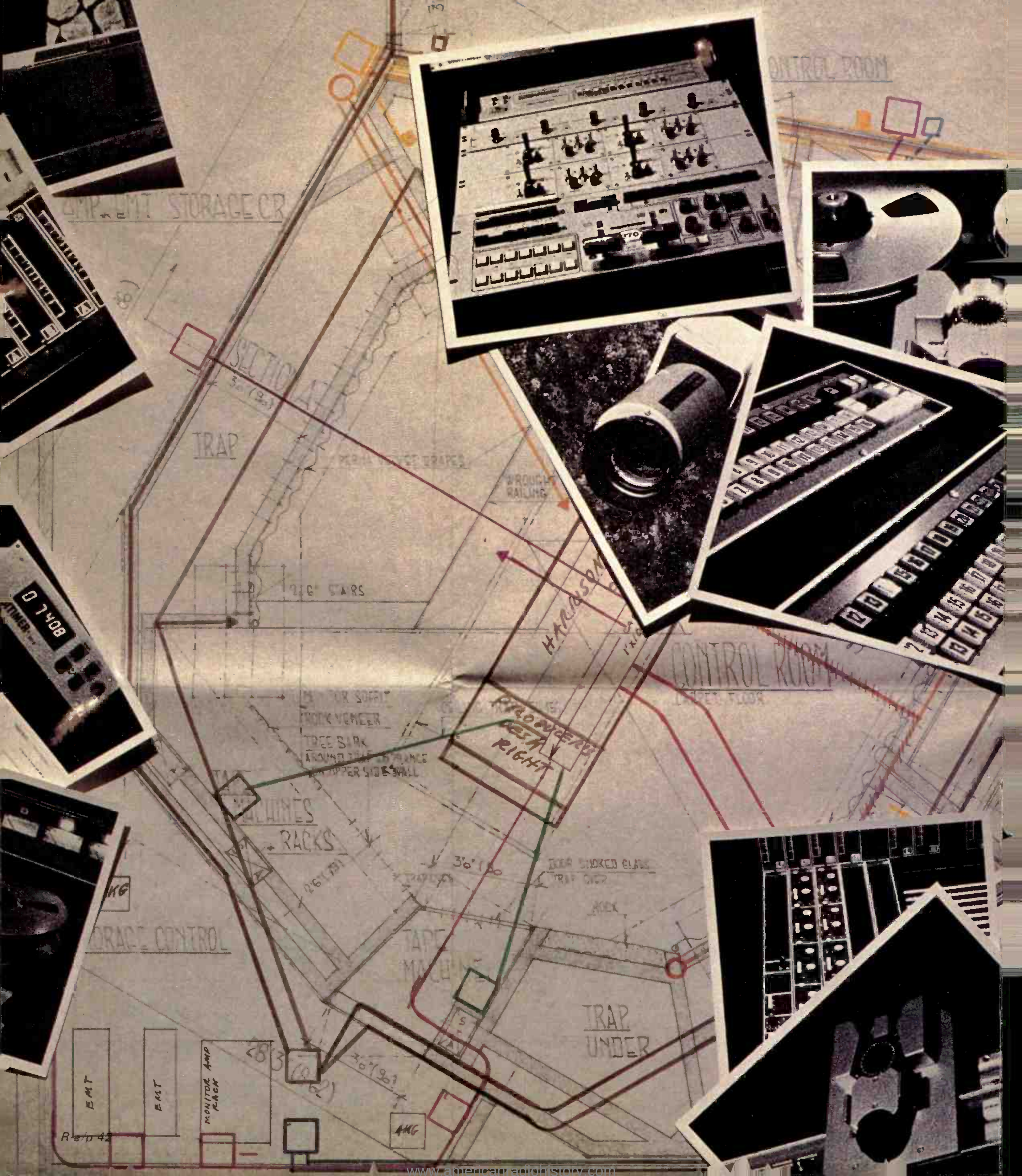
When sweetening, a synchronizer compares the two SMPTE time codes (VTR and audio machine) and locks the machines together. A cueing device enables the operator to run the machines back and forth from point A to point B while laying in various cues of music, FX, replacing dialogue from the overdub booth, etc.

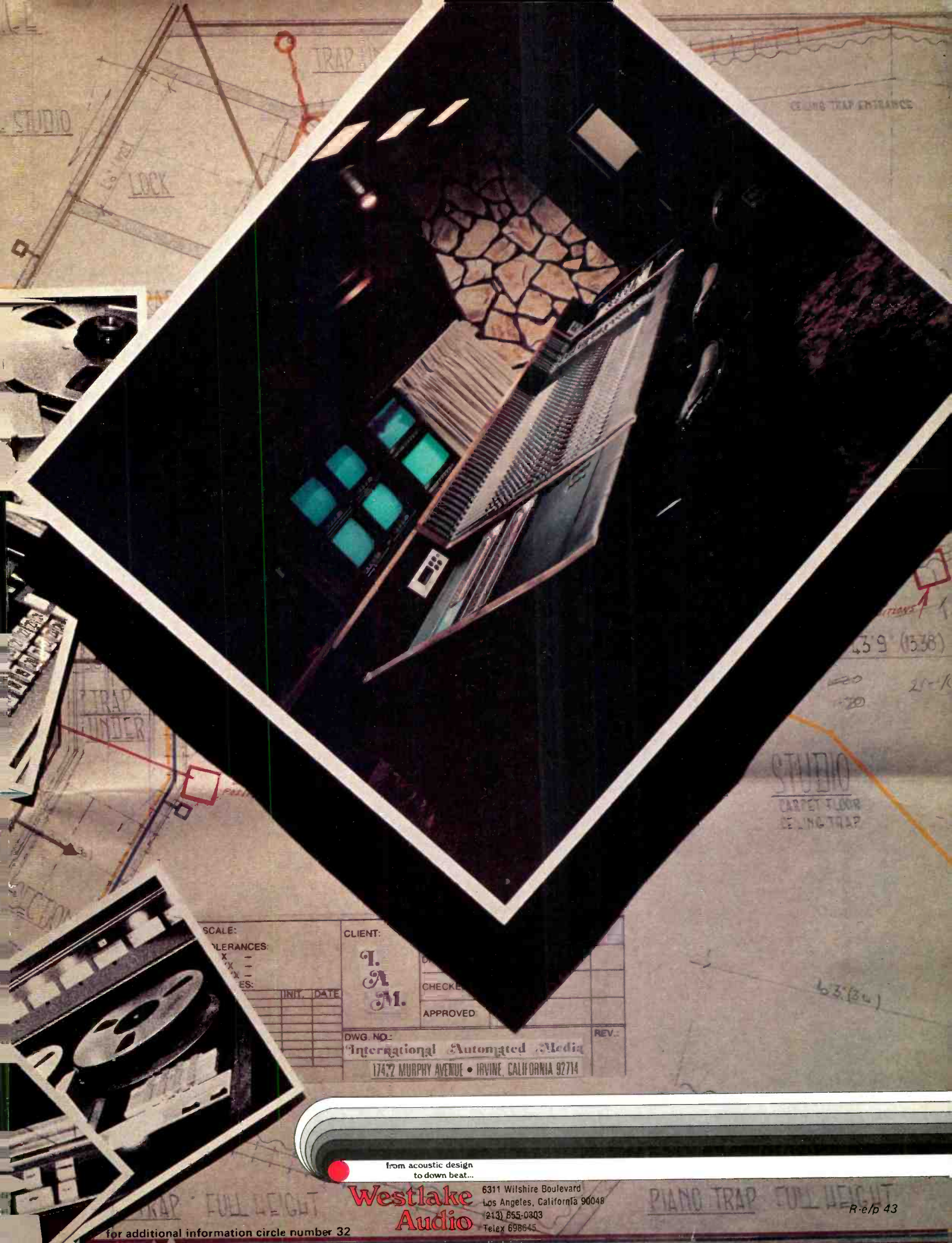
An editor commonly brings in music cues and FX to be laid in on the multi-track audio tape. The "transfer" box had to be thrown in to be consistent, since most editors would use an original rarely. Cartridge loops of opening or closing title numbers, FX, "fills", etc., may also be used when laying in on the multi-track, or they may be added directly at the time of the mixdown.

During the laying-in, or feeding process, a separate audio machine will be rolled at the appropriate time, the selected audio track on the multi-track machine previously being put into the record mode. The material on the separate, or feeding, machine is recorded onto the multi-track tape at the right place. The feeding machine might be cued "by hand", or a latching device of some sort might be used, wherein at a certain pre-programmed SMPTE code reading, the latching device would roll the feeding machine. Thus, a mark could be made on the feeding machine's tape, and the tape could be set on the mark each time. Two or three rolls might be necessary to get the timing right or to get the setting on the latching device correct, but with a little practice, the process can move along quite rapidly. (Some new equipment promises to make

. . . continued on page 44

A world of new ideas are found in Westlake Audio's second generation studios

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continued from page 41 . . .

this process easier and more precise.)

New or sweetened versions of musical numbers may be laid in on an open track, to replace former ones considered unsatisfactory. Hopefully, there will be a sync track of some kind to resolve if this is to be done, to avoid having to "hand slew" the number(s) with a VSO.

If audience effects are needed – laughs, applause, etc. – the man who specializes in this work will be called in with his "laugh machine" to lay down a track of such effects.

When everything needed has been laid down, the mixdown proceeds. There is no set format, and new procedures are developed from time to time. The author offers Figure 4 as an example of a medium adventure-type production he recently did. Track 2 was laid down at the same time as the individual master tracks – during the mix – via a common output buss assignment, to accommodate the chief engineer who wanted to have only one track to lay back, if possible. This redundancy would not be required otherwise. Two tracks of 2" videotape audio were laid over to avoid a re-lay should any drop-outs or other problems occur.

1	"Looped" Lines
2	Master Composite Mix
3	Production Track from Edited Video Master
4	Sweetened Production Track (pre-dubbed, includes looped lines
5	FX Track (No.1)
6	FX Track (No.2)
7	FX Track (No.3)
8	Music Track (No.1)
9	Music Track (No.2)
10	Production Track from Edited Video Master
11	Master Production Track (Dialogue) from Mix
12	Master Music Track from Mix
13	Master FX Track from Mix
14	
15	SMPTE Time Code
16	

FIGURE 4. TYPICAL AUDIO TRACK ASSIGNMENT FOR VIDEOTAPE POST-PRODUCTION SWEETENING AND MIXING

The sweetened production track (track number 4) is merely a copy of track 10, treated with EQ, reverb, filtering, including the looped lines, etc. This track could have been used as the Master Dialogue track if necessary. This particular show had audio tracks that needed separate attention prior to attempting to mix them in with the rest of the tracks that had been laid in on the multi-track tape.

A separate track was used in the mix for FX, Music, and Dialogue for the same reasons it is done in film; if a change is to be made, or if updating becomes necessary, one doesn't have to re-do the whole mix. Rather, the necessary track can be altered, and then the three can be combined in another lay-back. Also, foreign language versions were left open as a possibility, for which only FX and music would be wanted, the English dialogue being replaced by another language. This production did not require an audience effects track.

When the complexity of the production requires more than three FX tracks or two music tracks for laying-in, then a 24-track machine would be the logical choice.

Tracks 14 and 16 were avoided because of leakage of the SMPTE code and risk of edge damage to the code track.

LAY-BACK

When the mixdown is completed, the 16-track machine is synchronized with the master 2" videotape machine, and the new mix is laid down on the original master videotape, the old edited audio track being erased in the process. A completed master 2" show then exists.

DUBBING MASTER AND DUBBING COPIES

A copy is made of the complete sweetened 2" master videotape, which copy is used as a dubbing master to dub copies for the customers. The copies may be 2" copies, videocassettes, or whatever the customer requires.

As with the film diagram, some of the personnel shown might be the same person involved in a later step, such as a mixer, or a tape operator.

CONCLUSION

Perhaps we can draw a few general conclusions: The manpower requirements of tape are less than those of film. A half-hour tape show where no musical numbers are involved might take from 7 to 14 days to finish from the end of production shooting, whereas a half-hour film with no musical numbers might take 14 to 28 days from the end of production shooting to finish. Of course, a complex production or a few problems could add considerably to the time for either medium, and a crash deadline could shorten it. Videotape recorders and some other video gear are very expensive. But there are no laboratory costs involved in tape as there are in film.

Aesthetically speaking, tape sound is typically not as good as that of film, even though the technical ability of videotape audio exceeds that of optical film. Some reasons for this are:

–Tape is generally shot in less time than film.

–Tape productions generally have a smaller budget than film productions.

–You can get away with a lot over the small speakers in TV sets.

–Film sound is a finely tuned process where many experienced personnel contribute to create a polished end product. Trying to equal such quality in another medium is extremely difficult.

Nor does tape audio sweetening yet enjoy the type of control, speaking of the manipulation of the various elements, that hand editing of magnetic film allows. Consider the dialogue tracks that are built in film, for example.

Hopefully, television audio transmission and reception standards and practices will improve in future years. The PBS "network" for simulcasting stereo audio for the Metropolitan Opera Broadcasts is a hopeful trend.

The new helical scan VTR's with three audio tracks will add some possibilities from an audio standpoint, and EECO, working with Ampex, just introduced a new control and synchronization system that should be a big step forward toward better control of the various elements and machines in tape post audio sweetening.

It is interesing to note that at least one company on the West Coast is presently using its tape sweetening process in conjunction with other film equipment for film dubbing, and a few film shows have done their post-production sound via videotape sweetening.

This has been a healthy subject to tackle. May the Troll of Post-Production Sound forgive me if I left a step or two out of the dance or tromped rather rapidly over his bridge.

For the present, from an audio viewpoint:

– Nearly any type of production could be done on either film or tape.

– Some types of productions are cheaper and faster on videotape.

– Some types of productions are cheaper and faster on film.

– Tape audio is often (although certainly not always) not as well done as film sound.

– Any of the above are subject to change!

About The Author. . .
PAUL SHARP is a free lance mixer working with both film and videotape. After studying music at Brigham Young University, he worked in sound for the school's motion picture production department prior to relocating in the Los Angeles area.


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The Equalization Myth

. . . or, the importance of reverberation measurements in recording studio control rooms . . . . by Alan Fierstein

Monitor system equalization is the most widely used method of compensating for control room acoustics. Wtih a Real-Time Analyzer (RTA), the equalization process is fast, simple and cheap. Unfortunately, it is also generally wrong, because it overlooks the basic physical mechanism by which rooms affect the sound of a loudspeaker. This article will explain how a room affects sound and why real-time analysis is inappropriate. Then we will look at the proper means of correction as well as the legitimate use of equalization.

Imagine a room with smooth, hard, totally reflective surfaces. A sound introduced into this room would never die away; it would just keep bouncing around forever. In an anechoic chamber, however, sound is absorbed almost instantly (as soon as it hits the first highly absorbtive surface). These two rooms represent acoustical extremes, and in real rooms sound absorption take a finite time, and this time varies for different frequencies. A carpet-lined room would absorb high frequencies quickly but the low frequencies would be absorbed much more slowly. The way these reverberation times, or T_{60}'s* change at different frequencies is what distinguishes one room's sound from another's.

How do these frequency-dependent T_{60}'s affect the loudspeaker's sound? Well, first the sound emerges from the loudspeaker and reaches your ears directly. The sound then hits a surface which absorbs part of its energy according to the absorption curve of the surface material. A plywood panel will absorb more energy from the low frequencies than it will from the high frequencies that impinge upon it. Our carpeted room from before has just the opposite effect, of course. Lots of these reflections multiply this absorbtion characteristic many times and after the first sound has passed, our ears still hear the frequency modified reverberation. In the carpeted room we are left with a muddy sound, since the high-frequencies were absorbed quickly. Figure 1 shows the result of this heavily carpeted room. The initial sound consisted of two tones, a low and a high frequency of equal volume. In the balanced room this sound has decayed to a faithful miniturization of the original, but the heavily carpeted room has eaten up the highs and changed the spectrum from that of the original sound. Note that the high frequency wiggles are gone. We are left with a decayed low-frequency note only, hence the term "muddy sound". If you don't want a muddy reverberation, you must treat the room with materials that absorb low frequencies as quickly as high frequencies.

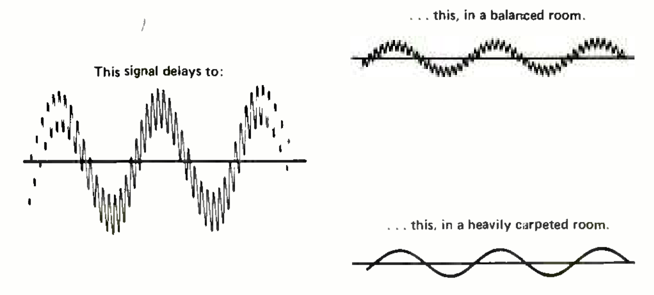

FIGURE 1

*T_{60} **is defined as the interval in which sound pressure decreases by 60 dB after a steady-state sound has been abruptly shut off.**

If we had been listening to music, our ears would have heard the new notes plus the muddy reverberation of past notes, giving the impression of added bass in the room. Can we avoid treating the room acoustically and simply equalize down the bass in the monitor system? No, because this equalization only affects the initial amplitude of the sound, it does not change the rate at which it decays. Figure 2 shows what happens when attempting to equalize problems like this. Please note that numbers and pictures are exaggerated here for clarity. In Figure 2a we see that the initial amplitude of the low and high frequencies are both 100 dB. The T_{60} of the balanced room is 1 second at all frequencies, so after 1 second both tones have dropped to 40 dB, which is essentially inaudible. Note that they both fell at the same rate from the same level and crossed the inaudibility threshold at the same time. In the heavily-carpeted room with the muddy reverb, Figure 2b, we have attempted to compensate by equalizing down the bass. The T_{60} of the bass is 2 seconds, and the T_{60} of the treble is 1 second. If we equalized down the bass 30 dB, it would start at an initial amplitude of 70 dB and fall 30 dB in the same time that the high frequencies would fall from 100 dB to 40 dB. Therefore, both tones would again become inaudible simultaneously. But we have made the reverberation tonal balance correct at one point only, at 40 dB, which is useless because since the decay times are different at low and high frequencies, the tonal balance is changing throughout the decay period. Also, the direct sound is now totally

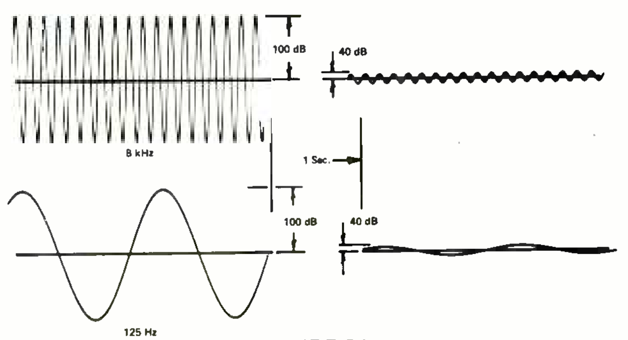

FIGURE 2A

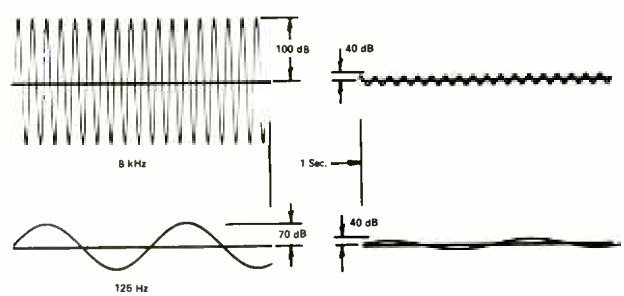

FIGURE 2B

non-flat. By contrast, the balanced room of Figure 2a has a flat direct sound, an unchanged tonal balance for the entire decay period and both frequencies reach inaudibility together. Clearly this is a much more desirable situation than the heavily carpeted, heavily equalized room of Figure 2b. The wonderful result of this balanced room is that a speaker that is flat in an anechoic chamber will sound flat at the mixer's ears, too, *without equalization*.

Contrary to popular opinion, a Real-Time Analyzer (RTA) does not display in real time, for if it did our poor slow eyes could not follow it. It integrates the input over a finite time period with a slow decay that makes observing reverberation impossible. On the RTA, the reverb of the room adds to the display of the pink noise, and a non-flat reverb characteristic will add more of some frequencies than others. For example, on the RTA our carpeted room with the muddy reverb will add low end to the display, giving the impression that the initial sound is bass heavy and that equalization is needed. The RTA's blind addition of signal and reverb is the root of the problem. RTA's are used with pink noise, which is a *static, continuous* sound, as compared with music and speech which are *impulsive* in nature. Impulse sound is defined by its initial level and time history,[1] and the RTA simply adds level and time history together in a way that our ears do not. Our ears hear the effects of room reverb during the pauses of music and speech. Pink noise has no such pauses.

How real is this effect in actual control rooms? Of course, reverberation 20 dB or more below initial levels will not add significantly to the curve height on a RTA, but the first 20 dB does. That the reverb is significant in affecting the RTA's display is born out by the fact that in a room with a T_{60} of .2 second, significant reverberant energy exists as close as 3 feet from the speaker. Obviously this depends upon other factors, most notably speaker Q. But when a speaker whose 1 foot frequency response of ±2 dB becomes ±12 dB at 8 feet (this actually occurred in a control room we measured) you can see that the room reflections have a pretty heavy influence. This wild response was not caused by standing waves. This room was plauged by a non-uniform T_{60} vs. Frequency curve. The ironic part of this story is that the speaker itself is obviously quite flat (±2 dB) and yet the room is giving this speaker a bad reputation (±12 dB). I wonder how many engineers are condeming their innocent speakers!

In addition to all this, equalizing the monitor system makes the important direct sound non-flat! Two rooms, equalized flat, can (and often do) sound different for this reason. Attempting to correct frequency-dependent time decays with initial amplitude equalization is like adding apples and oranges, and this basic error occurs regardless of whether you equalize to sine waves, pink noise, or "full-spectrum" pulses.

ROOM TREATMENT

Properly treating a room is a complex job. What follows is a synopsis of common problems and solutions and is not meant to be a do-it-yourself guide to an acoustics diploma. An experienced consultant is a wise decision if your room needs therapy.

Standing waves are a function of room dimensions and shape. Flutter echo is caused by multiple reflections between parallel surfaces. Room modes are room resonances that occur closely spaced in frequency and tend to reinforce their characteristic frequency when it is present in the program material. These problems are minimized by designing with few parallel surfaces, ensuring adequate diffusion and by isolating room resonant frequencies from each other by choosing optimum room dimension ratios. These are mentioned in reference 3. Speaker placement can also affect standing waves, I've been told.

Vibrating surfaces in a room can cause response problems in addition to the annoying "buzz" that is their most obvious manifestation. A surface that is free to vibrate can contribute to distortion of the acoustic field within the room that may be falsely blamed on speakers. This is sometimes called "whatnot" distortion and can generate harmonically related products in the tens of per cent. Another way that vibrating surfaces affect room response is by the manifestation of a response notch characteristic of a high Q filter. These notches show up equally well with RTA's and with sine wave sweep and reverberation measurements. The cure, of course, is to search out the vibrating body and stiffen it sufficiently to eliminate the vibration.

The symmetry of the speaker positions and the listening locations with respect to the symmetry of the room must be considered. Unless the gross problems outlined above are examined and corrected, no amount of equalization and reverberation analysis will solve them.

With the gross problems out of the way, the absorption is added, subtracted, or modified to provide the desired T_{60} in each frequency band, usually octave bands. This can be planned in advance to an extent by using tables of absorption coefficients that have been published for various building materials. You multiply the square footage of each material by its coefficient at each frequency, and then you add up the total for each frequency and apply this to a T_{60} equation such as the Norris-Eyring. But since no one has published the absorption coefficient of your console you'll need to take measurements of the T_{60} curve. Some may want a control room with a reverb curve approaching a typical living room's, or perhaps a flat T_{60} vs. Frequency curve is desired.

Finally, an equalizer can be used to fine tune the speaker system if its anechoic chamber response needs changing or if it was never tested in a chamber in the first place due to its custom design (often the case in studios). Usually the difference between one foot and eight foot frequency response curves points out the degree to which room reverb is playing a part, and here a RTA is handy.

To sum up, control rooms are not equalizers or filters (though they appear to be on a RTA screen), they are time-decay absorbers. Do not correct rooms with amplitude changes (equalization), correct their T_{60} curve instead. Equalizers are for fine tuning of speaker deficiencies that would show up in anechoic measurements, or for electrical modification of a recorded track, etc. When acoustical changes are not possible, as in many sound-reinforcement applications, equalization has the additional use of allowing increases of acoustic gain, if applied properly.

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about the author . . .
ALAN FIERSTEIN is the president of Acoustilog, Inc., manufacturers of reverberation measurement equipment. Previously, he served as maintenance engineer at Media Sound and Electric Lady studios in New York. He also designs, builds and maintains recording and film transfer operations in the New York area. He is owner, operator, and chief engineer of Sorcerer Sound, an eight-track studio in New York City.

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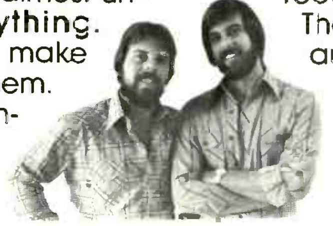

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psychoacoustics - part 1:

Time Delay in the Studio

by

Christopher Moore

PART ONE: THE BACKGROUND

In the seven years since Lexicon introduced the world's first digital audio delay unit, digital delays have become increasingly popular in modern recording studios. Now, more versatile units at reduced cost are available, and the future promises further advances. With all the delay units in use today, there is nevertheless a shortage of conveniently available information to help the recording engineer use this powerful tool. This two-part article is intended to provide a background against which the basic applications of time delay may be presented, and to form a basis from which the recording engineer may go on to creatively develop new uses.

In part one, basic concepts of room acoustics are reviewed, covering the important ideas of direct sound, early reflections, and reverberation. The particular emphasis, of course, is the role of time delay. We continue with a brief presentation of some relevant aspects of psychoacoustics, including an explantion of how we localize sounds and how time delay enters into these processes.

In part two, we will present a "cookbook" of techniques and effects that can be realized in the studio today, with present equipment. This is intended as basic, hands-on application material lightly sprinkled with theory. Many mixdown effects will be described, and several performance and live recording uses discussed.

Why Time Delay?

Since the advent of electrical sound recording, music has become increasingly accessible to most people. By now it is safe to say that the greatest majority of all music is heard from recordings, and not directly from musicians. However, in spite of the predominance of recorded music, our hearing is conditioned by live experiences. We do still hear live musicians, thank God, and we hear most of the sounds of our daily lives in a variety of acoustic spaces. And even the recordings we play are undeniably shaped by the acoustics of our homes. All this real-life hearing has strongly and irrevocably shaped our perception of sound.

We hear with two ears, most of us, giving rise to a range of significant capabilities. The sound at each eardrum is nearly always different; some differences are due to intensity or frequency response discrepancies, while others are due to time/phase differences. These differences are essential in determining the apparent position of the source and are important in giving us a sense of the acoustic space surrounding the source and the listener.

The intention of recording engineers varies widely, from attempting to faithfully translate a live listening event to the consumer's home, to creating a new and exciting sound with no reference to a familiar acoustic space. Whatever the intent, however, the recording will be heard by mortals such as ourselves who have two ears and who have been experiencing their world through them for as many years as they care to remember. So it seems important to include at least a few clues in the recording that will give us a sense of familiarity, at-homeness, of the sound being rooted in some sort of acoustic space, or ambience. Engineers acknowledge this need by using pan-pots, reverberation, multi-mike recording, and delay units, as well as other, more specialized devices to create two or four channel recordings.

Time delay units have come to be used to great advantage in today's world of highly separated multi-track recordings. These devices can restore a sense of ambience and spaciousness that will root newer musical styles and forms in a famil-

ABOUT THE AUTHOR:

Christopher Moore is a project engineer in Lexicon's digital audio engineering department. His long industry experience and work on applications for delay based effects at Lexicon provided the basis for this two-art article.

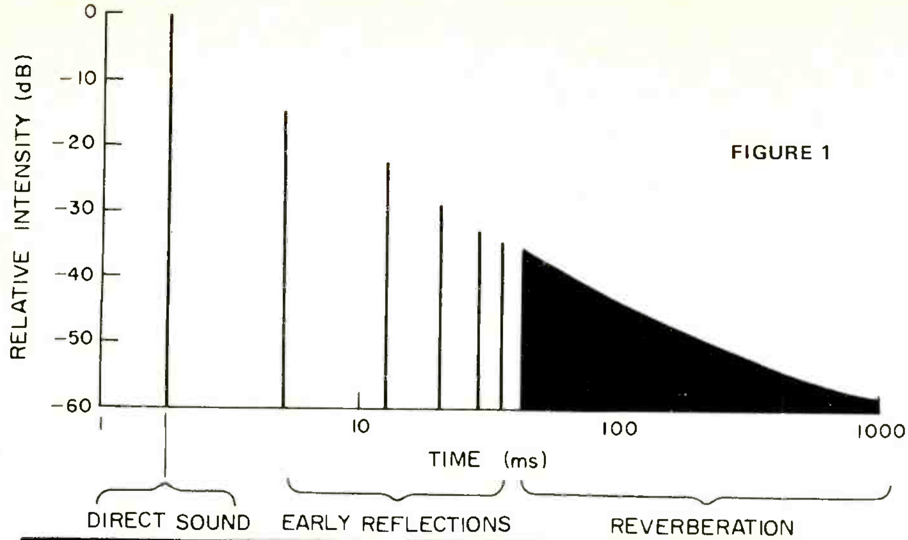


FIGURE 1

iar, common ground of acoustic space. In addition, digital delay can create a variety of special effects and enhance the sound of single performers or small ensembles.

Room Acoustics

When we hear a musical performance live, we are generally blissfully unaware of the variety of sounds that blend together into our overall perception. We are aware of the musicians themselves and their location; we have a sense of how large the room is and where we are in it; and we can hear how reflective the walls are. How do we come to know these things? To be sure, we are vastly aided by our sight, but even blindfolded we could perceive most of these aspects of the room. It is convenient first to divide our perception of sound in a room into three time intervals (Figure 1). Of course, the intervals are not sharply defined for every combination of source, room, and observer, but rather blend into a continuously perceived experience unless the acoustics are rather poor.

Direct Sound

The first sound we hear is the direct sound which comes to us by line of sight from the source (Figure 2). The direct sound is extremely important, as it arrives pristine and uncluttered by whatever derivative sounds the room acoustics will produce a moment later. The direct sound, of course, lasts only as long as the source emits it. Now, although we will fuse this sound with the early reflections that follow, it still has paramount importance due to the psychoacoustic phenomenon called the precedence (or Hass) effect. This causes us to suppress the later reflections and use the first sound to determine the source location, source width, and the disposition of individual instruments. In addition, the direct sound comes to us with a different frequency response than the later sound, a frequency response truer to the sound produced by the source because it is subject only to high frequency air absoption. The all-important transient characteristics are conveyed by this direct sound, as is timbre, especially in the high registers.

Early Reflections

The second time interval of our sound perception begins with the arrival of the first reflection from a room boundary. This will be followed by other reflections, increasing quickly in time density while decreasing in amplitude. These reflections are collectively called the early sound and are due to sound emitted by the source in directions other than the direct line toward the observer. These sounds reflect in the manner of rays from the room boundaries, sometimes bouncing off only one before reaching us, and then off of several boundaries in succession (Figure 3). Each reflection involves loss of energy due to absorption by the boundary, so that the early sound not only is attenuated, but is altered in frequency response. The wall coverings determine the degree of absorption; in this way the character of the early sound can tell us in a qualitative way how hard the walls are, how 'live' the room is.

The spacing in time of early reflections is related to the path lengths, and therefore the room size. In an unconscious way, our ear/brain combination will process these delay times into a sense of the room size. An opening to another room, or to open air, will provide a hole through which otherwise reflected sound will leave the room, and this can be sensed as well.

The early sound fuses with the direct sound due to the precedence effect and modifies it, making it louder and more spatially diffuse, and altering its timbre. In this way, the early sound becomes an aspect of the source itself, giving it warmth, body, and ambience.

Reverberation

In a short while, the time and spatial densities of the reflections increase to the point where individual reflections are lost in the shuffle and we are in the last time period, the reverberation time. The reflections excited by the already silent source are coming from all directions (Figure 4) and so frequently (Figure 1) that they blend together to form a diffuse, gradually decaying reverberation that envelopes the original sound with its warmth and body. If we consider that most sources don't tend to beam their sound in

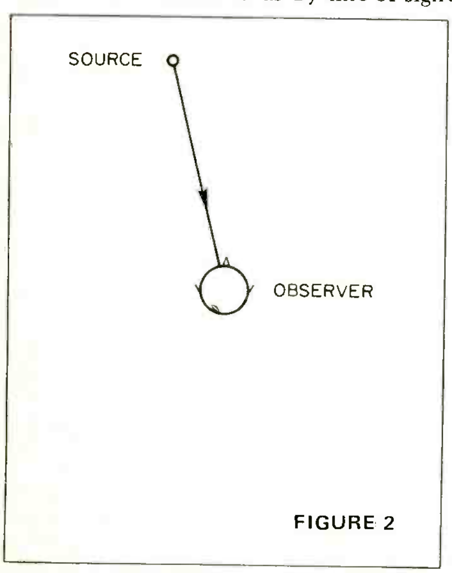


FIGURE 2

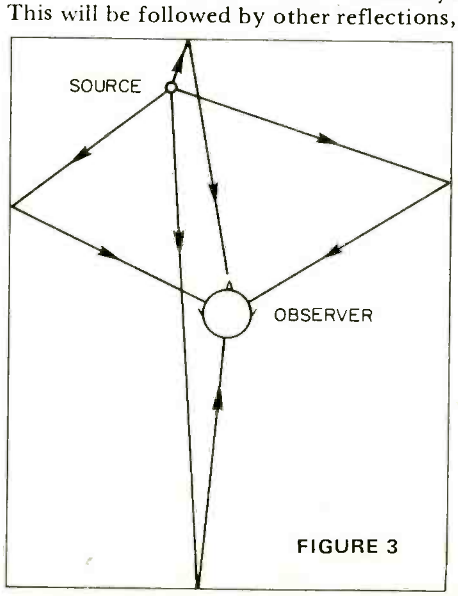
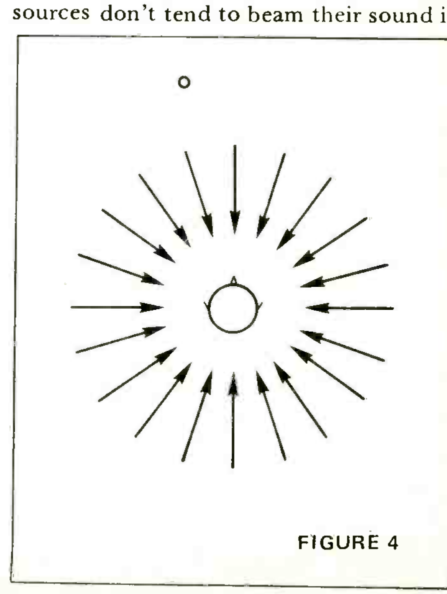


FIGURE 3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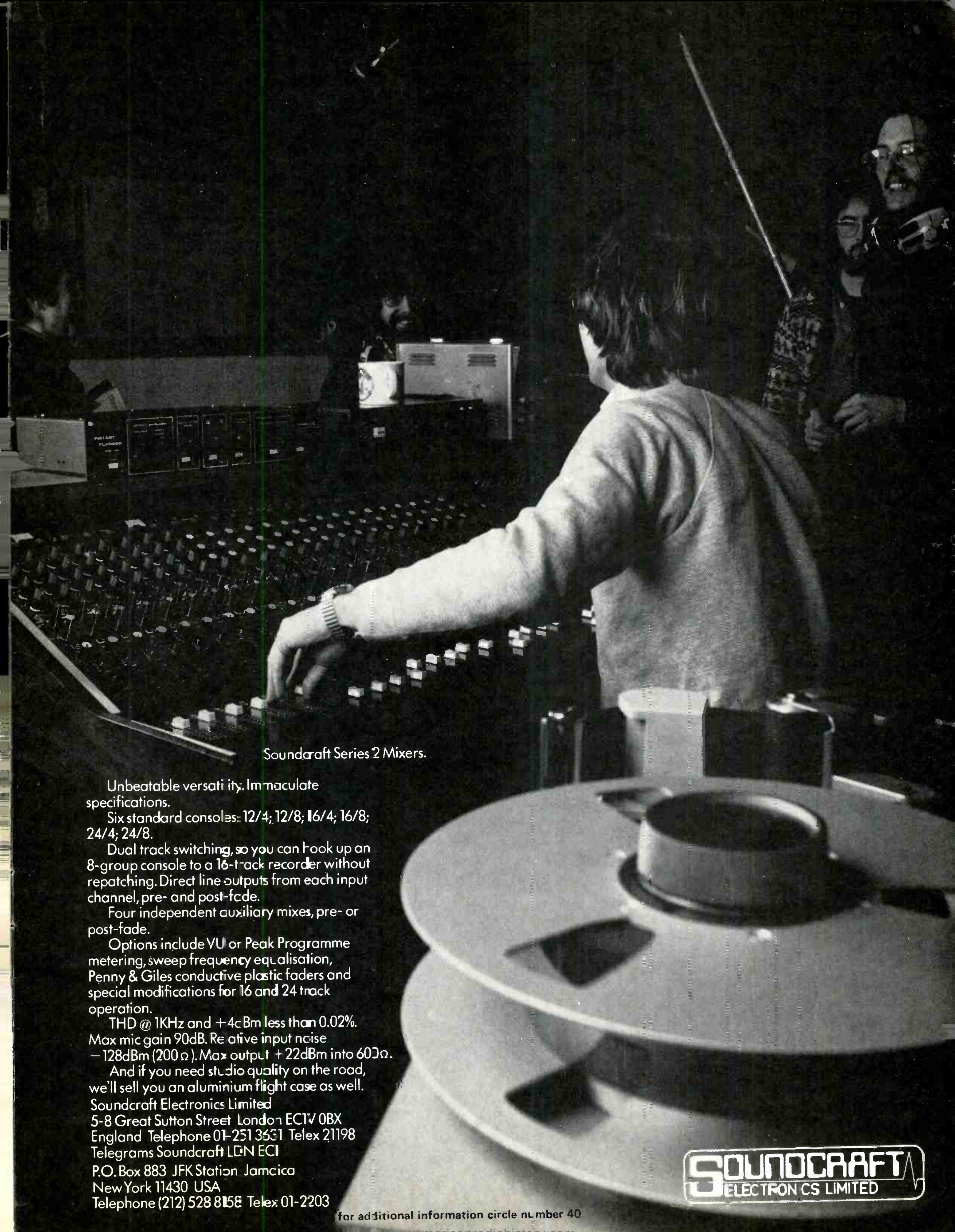

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a narrow path, but radiate acoustic energy into the room in a large three-dimensional angle, then it's clear that little of it reaches us as direct sound. The bulk of the source energy goes off in all directions and excites the room's resonant modes producing reverberation. Reverberation is characterized by its spectrum, which is generally like a low pass filter, attenuating high frequencies due to wall and air absorption, and enhancing the lower frequencies. The time it takes for the reverberation to build up after the source emits the excitation is related to the room size, as is the so-called decay time (the time for reverberation to die down to 60 dB below its maximum level).

Reverberation helps modify our perception of sound in several ways. It reinforces the source, surrounding it with a warm mantle of ambience and increased volume. Reverberation also aids us in estimating the distance to the source. Consider that the bulk of the source energy ends up as reverberation. This means that unless we are very near the source, most of the sound we hear will be from reverberation. The ratio of direct to reverberant sound, then, is a function of our nearness to the source and is used psychoacoustically to estimate distance. Furthermore, the nearness of room boundaries, the presence of openings, and the degree of boundary absorption all affect reverberation, so that we get clues about these aspects of the room from the reverberation. Reverberation can also work against us, especially if there is so much of it, lasting for so long, that it provides confusing echoes that linger and interfere with the intelligibility of the source, making it hard to hear short transients and burying the high frequencies in cloying, muddy overhang.

In summary, when a musical source in a room produces a sound, some of its energy travels straight to us and arrives first as the direct sound. From direct sound, we learn the nature of the source, its location, and its spatial distribution. Next, the early sound, produced by reflections of the source energy from room boundaries, arrives from other directions. Although the early reflections blend smoothly with the direct sound, enhancing its loudness and body, they also define the room size by their delay. Finally, some time after the direct sound, the room truly begins to speak from all directions, delivering the great bulk of source energy as reverberation. This sound, spatially diffuse and impossible to localize, blends with the source also, but provides an essential impression of the room liveness, adding color and body to the sound source before it dies away.

The kind of music, instruments employed, and how they're played determine how these three aspects are perceived. Brief, impulsive sounds and sounds with sharp initial attack reveal the most about the source location, room size, and reverberant qualities. Muted instruments, legato passages, and instruments with fewer transients blend in with the early sound and reverberation making them harder to perceive and utilize psychoacoustically.

Having reviewed something of how rooms work, we move on to see how our hearing uses these sounds to deduce the source location and room characteristics. It will be seen that although the direct sound is most important, each time period discussed has its psychoacoustic uses.

PSYCHOACOUSTICS

The significant topics from psychoacoustics included here are: the localization of a single source, the localization of multiple spatially separated but coherent sound sources (precedence effect), and the perception of multiply repeated sounds from the same location (temporal fusion).

Localization of a Single Source

First of all, what do we mean by localization? To locate a souce in three-dimensional space, we must establish three quantities (three-dimensional vector space). One possible set of three measures includes the direction, defined by an angle to the source projected onto an imaginary plane parallel to the ground intersecting our head at ear level (the lateral plane). This establishes the front-back, left-right placement of the source (see Figure 5). The other two measures are elevation (angle above or below the lateral plane) and distance. Since time delay is only secondarily useful in influencing our perception of elevation and distance, they won't be treated here.

To localize a sound, our ear/brain mechanism relies most heavily upon the direct sound and works with interaural amplitude differences (IAD), interaural time differences (ITD), and interaural spectral differences. Interaural difference means the difference between the signals present at each eardrum. Generally, the two acoustic pressure waveforms will have much in common, but it is their differences that are most useful in binaural perception of location.

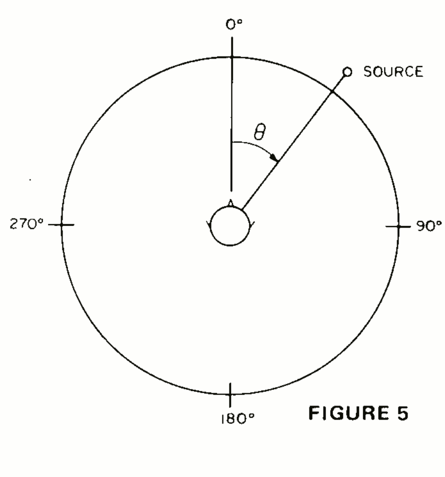

FIGURE 5

Interaural Time Differences

The direct sound from a source reaches both of our ears, but the physical separation of our ears by eight to nine inches results in different path lengths causing one or the other ear to receive the sound first (Figure 6). This will be true for any source not on the median plane. The physical spacing is the basis for Interaural Time Differences (ITD) in our hearing. The time delay difference is a function of the direction from which the sound arrives. The delay varies from zero for sources at 0 degrees (directly in front, on the median plane) to a maximum of about .8 ms for sources directly opposite the left ear. Our hearing mechanism uses this ITD to help sort out the direction.

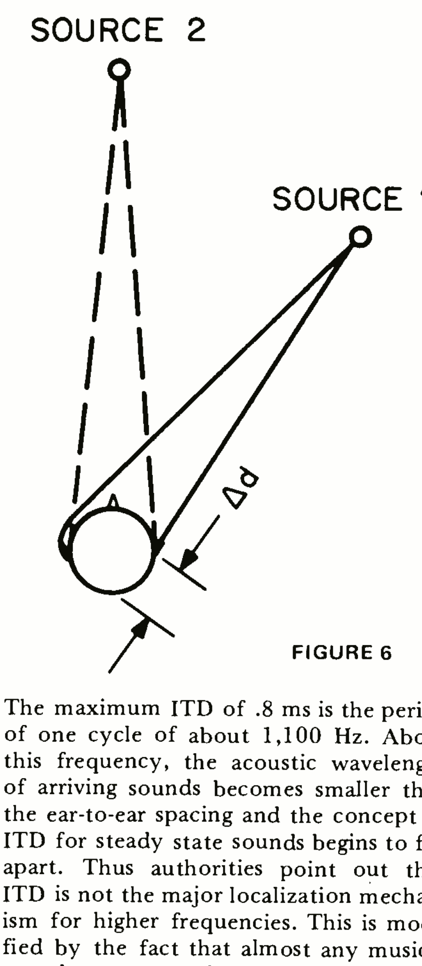

FIGURE 6

The maximum ITD of .8 ms is the period of one cycle of about 1,100 Hz. Above this frequency, the acoustic wavelength of arriving sounds becomes smaller than the ear-to-ear spacing and the concept of ITD for steady state sounds begins to fall apart. Thus authorities point out that ITD is not the major localization mechanism for higher frequencies. This is modified by the fact that almost any musical sound, even a steady state one, has some transients (attack envelope occurring when changing notes for example) with energy in a broad spectrum that develops ITD.

Precedence Effect

So far we've dealt with localization of a single source. We move on to consider the interesting cases where roughly the same sound is heard originating from more than one location. This is a common occurrence in sound reinforcement, where both the real source and the loudspeakers can be heard in some seats. The two or four speakers at home in a hi-fi system are also multiple sources of the

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same or similar sounds. But there don't have to be two or more actual sources for this situation to arise. The early reflections already discussed come from different locations and are similar to the actual source, so we must include this instance as well.

Researchers, notably Haas, have given us a well-accepted theory explaining how we perceive these multiply positioned sources: we tend to localize a sound at a position from which the first arriving sound originates. Within a range of time delays, from about 5 ms to 30 ms, the effect is so strong that we must increase the delayed source amplitude by over 8 to 12 dB before IAD can pull the signal back to center. In Figure 7 we show two sources and two observer positons.

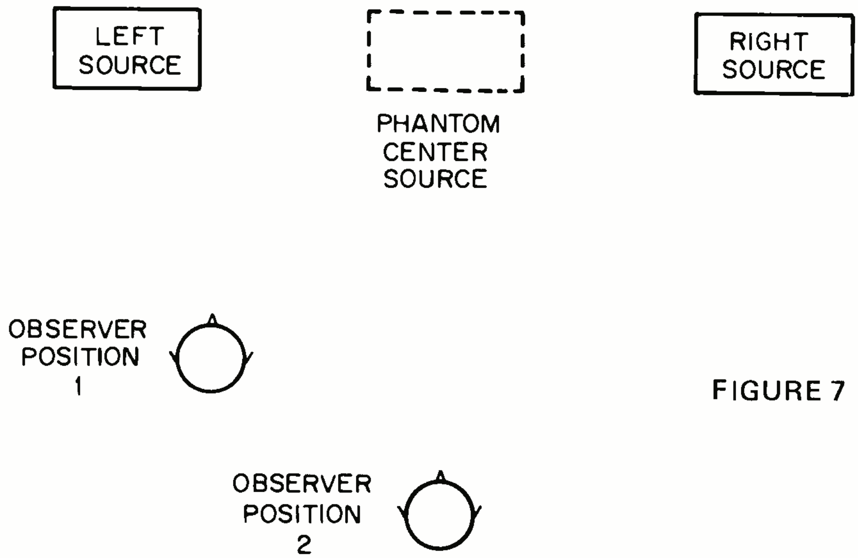


FIGURE 7

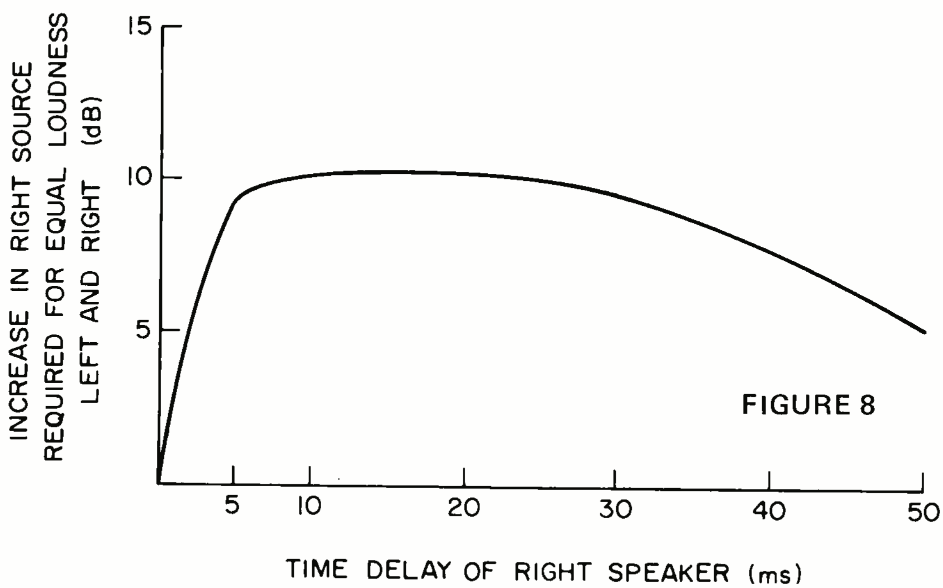


FIGURE 8

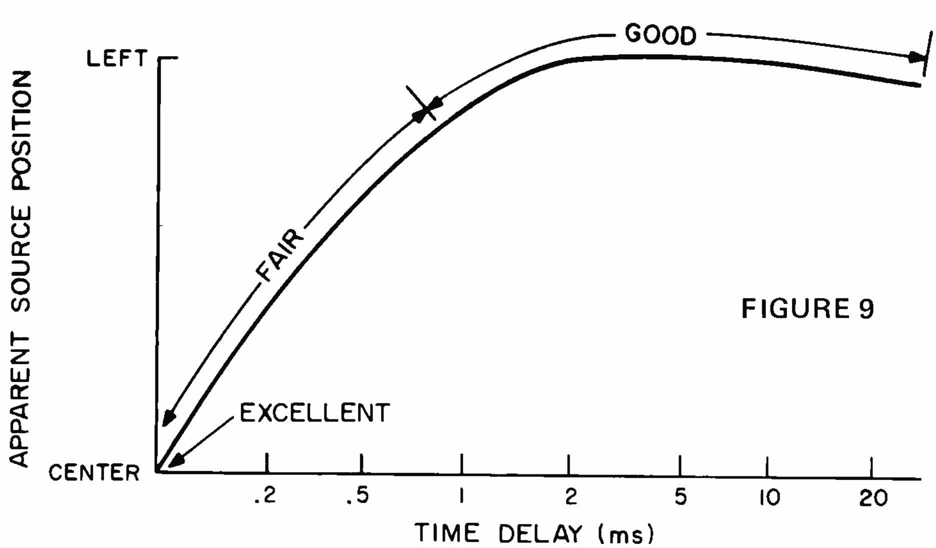


FIGURE 9

Consider for a moment that Source L and R are identical. For a forward-facing observer located anywhere on the plane of symmetry midway between the two speakers, as at Position 2, there will be no IAD or ITD (both eardrums receive identical signals, at least from the direct sound). So we perceive a single phantom image midway between the two speakers. If we move closer to L (position 1) its signal will arrive earlier producing the familiar effect where that speaker takes over and we are almost unaware of Speaker R. Now if we go back to the center position and introduce a delay in the electrical signal fed to the right speaker (finally we come to digital delay), the source will appear to come from the left, even though each is emitting the same acoustical power. Localization is, as we've seen, dependent upon both IAD and IDT, thus it comes as no surprise that we can influence the apparent source direction in the set-up of Figure 7 with both amplitude and time delay differences. This is seen in Figure 8 which shows the increased amplitude needed (vertical scale) in the delayed speaker to offset the localizing effect of the delay and maintain the impression of equal loudness from each speaker.

Figure 8 shouldn't be taken as gospel, since these effects are rather dependent upon the spectral and dynamic characteristics of the signal. Steady state sounds are affected less by delay differences, since by its very nature delay detection requires some event (i.e., a transient) to serve as a reference. Another useful curve is given in Figure 9 and shows the apparent source position for equal loudness sounds as a function of delay. Note the non-linear nature of this curve: very rapid shifts of position occur for small delay differences up to about 2 ms. From 2 ms up, the image becomes better and better defined at the non-delayed speaker until around 30 ms (again, dependent upon the program material), when the delayed signal begins to break apart into a distinguishable echo, at which point the concept of localization no longer applies, because the two sound sources no longer fuse into one. This idea of fusion brings us to our last important topic from psychoacoustics: the fusion of sounds occurring separately in time and/or space into a single sound.

Temporal Fusion

Actually, temporal fusion is really inextricably linked to the precedence effect already discussed, since we assumed that the two sounds differing in position or arrival time were fused into a single sound that was then localized. It seems that our ear-brain apparatus has learned to blend together two or more very similar sounds arriving at nearly the same time. After the first sound is heard, it is as if the brain suppresses anything similar arriving within about the next 30 ms. In this way, the sounds are heard as one and intelligibility is greatly enhanced. Beyond about 30 ms, but depending on the nature of the sound, the sources become individually distinguishable. The time limit of 30 ms is, again, not a constant, being nearer to 4 ms for single, short duration clicks, and over 80 ms for slowly changing sounds.

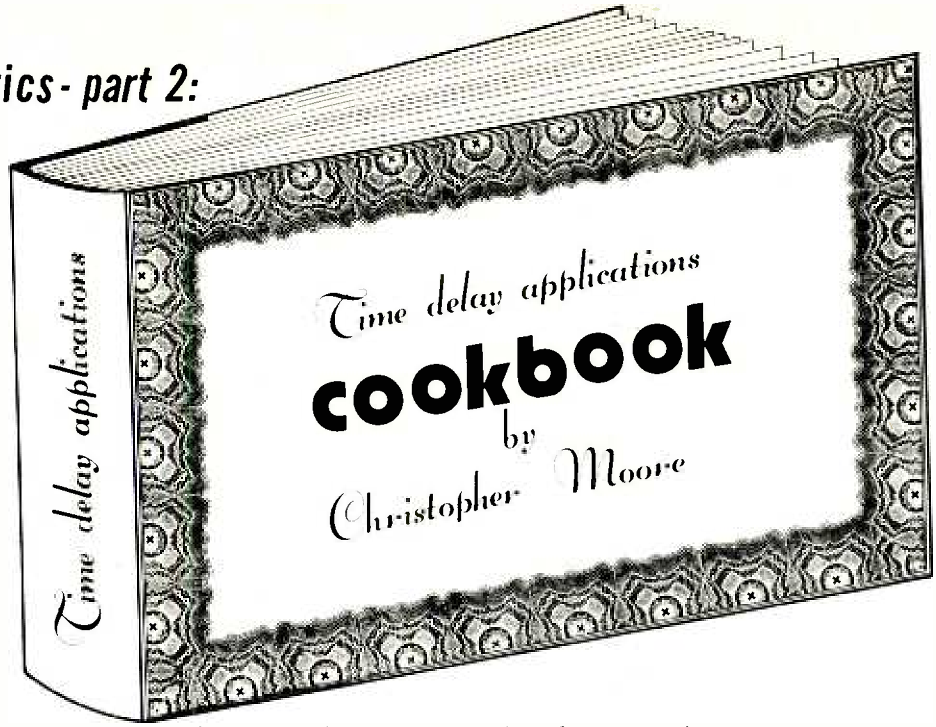
It is fusion that permits us to blend all the early reflections together so that they merge with the direct sound into a new, louder, fuller sound. This last idea is important – namely, that although we hear only one sound, it seems to be louder, or intensified. This is a useful phenomenon easily achieved with digital delay. If the sounds originate from the same point, we hear them as one sound, intensified, from the common location. If they come from different locations, we again hear one sound, but a louder one, coming from the direction of the earlier source. Temporal fusion works as long as the gaps between arriving signals are less than 30 ms or so, and provided that later arrivals aren't much louder than earlier ones. It is this phenomenon that keeps the direct sound, the early reflections, and reverberation all together as one pleasing and natural perception of live listening. Remember, though, that our directional hearing mechanisms work on the direct sound, especially its transient aspects, permitting us to localize the sound even though it is followed within milliseconds by multiply occurring reflections from different directions, and by reverberation 'echoes' so closely spaced in time, and so scattered in space as to be completely unlocalizable.

PSYCHOACOUSTICS AND TIME DELAY IN THE STUDIO

Before leaving this background material article, let's pull together the things discussed so far and see how they'll be useful. In reviewing room acoustics, we saw that the direct sound from a source was localized both by IAD and ITD. Since the ultimate product of a studio is usually heard on two speakers, we can use time delay as well as pan pots to help localize, broaden, and intensify a signal by sending the same signal, delayed, to other speakers. Further, since direct sounds are followed by early reflections, we can greatly enhance a mono-miked source by mixing several delays into both (or four, if quad) channels. These delays will intensify the sound and root it in an ambience otherwise hard to obtain. Now, since a reverberant room doesn't begin to 'speak' right away, we can delay the echo send signal and make the reverb sound more natural as well. Since the temporal fusion effect reassures us that we can add a delayed sound to the original and only experience a loudness intensification, we can use delay to increase the perceived loudness without increasing the peak or VU meter level. Two or more musicians playing the same line, in unison, provide a fuller, richer sound even though we hear only the merged, fused sound of their spatially distributed voices or instruments. Delay can be used to simulate this fuller sound by processing a single performer, or a small group of performers with additional time delays (this is the so-called 'doubling' or 'double-tracking' effect). The additional time delays can be stereo-panned to produce the effect of a larger number of performers spread out between the speakers, or mono-panned bunching them together in lateral direction, but with greater depth.

Other uses of delay will be given that have a less natural basis in room acoustics or hearing, but which are nonetheless exciting and valid. In any event, if you've read through these background sections so far, I hope they will serve as food for thought and future discovery as well.

psychoacoustics - part 2:



The basic element in all these applications is some sort of signal – mono, stereo or multiple track – whose sound we would like to improve. We must have a device for achieving time delay with at least one input channel and one output, although some of the effects will require additional capability, such as two channels of delay, multiple outputs per delay channel, continuously variable delay over a limited range, and VCO modulation of the time base. The Lexicon Delta-T 102-S is a particularly versatile delay unit that provides all this capability, including VCO.

There are a number of ways to use a digital delay for signal processing. Perhaps the most important uses are to simply obtain delayed versions of a musical signal that can be mixed into the final stereo or quad recording. This is rather natural signal processing, not altering the quality of the sound in any radical way. In another class of delay processing, the delayed outputs are coupled back to the delay input, causing feedback, either with long delay times, producing echoes, or with shorter times, resulting in tuned filter or ringing effects. Using a VCO to vary or modulate the time base opens up other kinds of signal processing that may in fact transform the track into a new and almost unrecognizable sound. Our approach will be to proceed from the simple, most natural effects to the most dramatic ones.

Image Placement

In the first part of our article, we saw how to localize a source in a real listening situation. We can apply some of these ideas at mixdown to place instruments in different apparent locations. The pan pot is widely used to determine apparent source direction, but delay is also a viable method and has certain advantages. A panned image is localized at or near a speaker by amplitude differences, seeming to come from the speaker with the loudest signal for a listener in an appropriate central area (the so-called 'stereo seat'). But as soon as the listener moves out of the stereo seat, nearer to one speaker, it takes over as the apparent source due to the Haas effect, even if it's the one with the weaker signal. The Haas effect therefore makes this form of image placement less stable for varying listener location.

Figures 7 and 8 in Part One show that a time delay difference of from 5 to 30 ms will firmly locate source direction at the earliest arriving speaker in spite of the two speakers emitting equal loudness signals. So by sending the zero delay signal to the desired channel direction, and delayed signals to the others, we ensure that the signal appears to come from the chosen direction for listeners in most locations of the room. Furthermore, the sense of ambience and loudness will be increased due to the supporting delays arriving within the temporal fusion time. This technique is especially useful in quad, where we can firmly establish a frontal source position (where it belongs, since we're used to facing musicians) by feeding the back channels with delayed signals. The delayed signals don't detract from the front direction established by the zero delayed signals and add a sense of the ambience of a room speaking from behind. Except in rare cases where the producer wants to assault his audience with surrounding musicians, this is a far more natural and satisfying way to present a musical experience. Figures 1 and 2 show typical delay times and channel assignments for this effect. Although shorter delays will also establish direction, they may lead to undersirable comb filter effects in the stereo or mono versions of a quad mix. Audition in mono during mixdown to be sure what's going to happen. In most cases, with delays over 15 ms or so, the effect is likely to be pleasing doubling or loudness intensification and there'll be no problem.

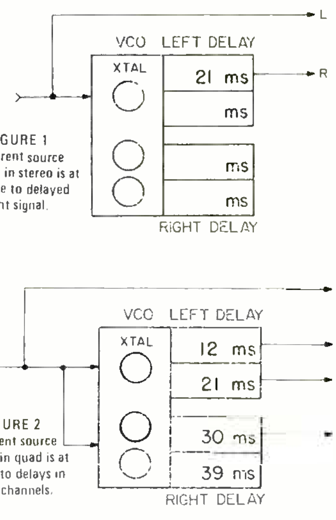


FIGURE 1
Apparent source position in stereo is at left due to delayed right signal.

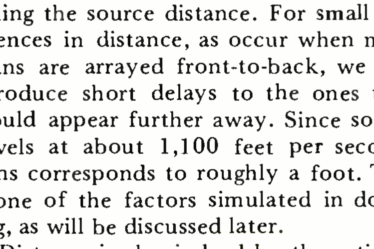


FIGURE 2
Apparent source position in quad is at LF due to delays in other channels.

Image placement also involves establishing the source distance. For small differences in distance, as occur when musicians are arrayed front-to-back, we can introduce short delays to the ones that should appear further away. Since sound travels at about 1,100 feet per second, 1 ms corresponds to roughly a foot. This is one of the factors simulated in doubling, as will be discussed later.

Distance is also judged by the ratio of direct to reverberant sound. A source that should appear to come from a great distance should then not appear in the mix, but should be delayed and fed to a reverberation chamber whose outputs (preferably one per mixdown channel) are fed into the final mix. In this way we create the effect of a distant source whose sound takes awhile to arrive and excite the room reverberation, and whose direct sound is lost in the reverberant sound of the room.

Doubling

Doubling is a technique that combines original and delayed signals in order to make a single performer sound like two or more, or a small group sound larger. It is also used to make tracks appear louder, wider, deeper, and more rooted in a room ambience.

The sound of multiple performers will differ from a single performer in a number of ways. For one thing, the sounds will arrive at our ears at different times. A second performer will not be in the same place, but will be in front or behind, left or right, causing his sound to arrive at different times from the first performer. Even if the two were in the same spot, they wouldn't play exactly in sync, but would wander slightly in time relative to each other, producing changing attacks that we unconsciously decode into the sense of multiple performers. Finally, the two or more performers will not play or sing exactly on pitch with each other. This, too, is part of the sound that tells us there are multiple performers.

How can we create multiple performers given a track of only one? Panning a delayed version into the same direction as the original will simulate a second performer behind the first. Panning a delayed version to another position will simulate a second performer in a different direction. Mixing in a delayed version with subtle pitch and time shifts will more naturally simulate a second performer playing with the differences in timing and pitch that we associate with human performers. To do the latter, a VCO function that modulates the digital delay time base is needed.

Properly done, doubling will not only give the impression of a multiplicity of performers, but will also increase the apparent loudness by more than the corresponding increase shown on a VU meter. This is called the loudness enhancement or intensification effect and is due to the temporal fusion that occurs with delays less than 30 to 50 ms. In doubling with more than two delay taps, it's possible to use delays greater than 50 ms as long as taps are spaced in between so that there are no delay increments greater than 25 ms or so. Longer delay increments are to be avoided, especially if panned away from the zero delay original signal, since spatial separation permits the delayed signal to break away into a perceivable echo more easily. The relative level of delay returns in doubling must be adjusted by ear, usually to no more than unity gain relative to the zero delay signal. If too loud, they tend to take over, turning the original signal into a pre-echo; at unity gain, the returns will sound like a second equally loud doubled source; and at reduced gain, the doubling voice will be more easily masked and may resemble early reflections more than another performer. Doubling can be used with all pop music instruments, although voice, guitar, piano, drums, and strings are the most common.

The basic signal flow chart for a simple mono doubling is shown in Figure 3. The

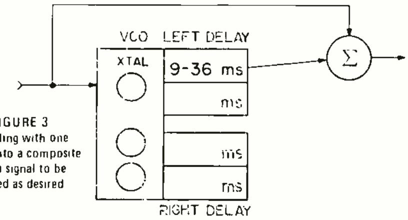


FIGURE 3
Doubling with one delay, into a composite mono signal to be panned as desired

mono source and one delay tap (although more than one can be used) are mixed together into a new mono composite signal that can then be positioned as usual with pan pot to the desired direction. The delays must not be too short (less than 10 ms for most programs) or comb filtering will be audible, and the delay differential should be kept under 25 or 30 ms unless the source is extremely smooth and legato. Since signal and delays will emanate from the same direction with this mix, the room acoustic event simulated is that of multiple sources at increasing distances behind the primary one, or of early reflections arriving after bouncing off the wall behind the source. This kind of doubling with one or two taps at 25 to 55 ms, is good with a snare or kick drum.

To our ears, however, it's nearly always more pleasing to use the doubling approach shown in Figure 4 or 5. Once again, the mono signal is fed to the delay unit and one or more taps brought back at appropriate delay settings and amplitudes, but now the original signal and delays are panned individually. For example, using four delays, the mix of Figure 5

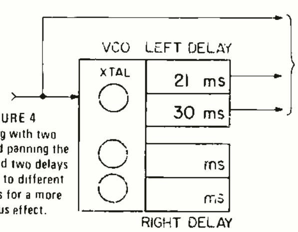


FIGURE 4
Doubling with two delays, and panning the original and two delays as desired to different directions for a more spacious effect.

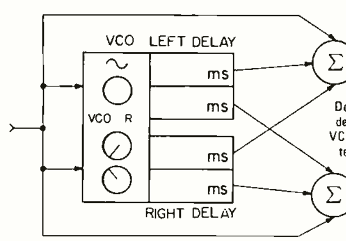


FIGURE 5
Doubling, with two delay channels and VCO, into stereo. See text for suggested delay settings.

would produce a strong mono-panned center image, nicely supported by the four delays criss-crossing in increasing increments left and right.

This will add ambience, apparent loudness, and the impression of additional performers arrayed across the stereo stage. This should be done with a stereo delay line whose channels can be individually clocked, so that the otherwise identical right delay channel has its time base under slow modulation by the VCO. This causes the right channel delays to vary above their settings and introduces a slight pitch modulation. A one-cycle-per-five-second (.2 Hz) sine wave is effective for this.

Suggested ballpark delay settings for Figure 5 are, from top to bottom: vocal, 15, 30, 45, 54 ms; string section, 6, 15, 30, 51 ms; kick drum or piano, 24, 45, 60, and 84 ms; and acoustic guitar, 15, 30, 42, and a teeny bit even of 105 ms.

This 'stereo doubling' is especially good with sources that we'd expect to be spatially distributed anyway, such as a string section. The effect has been reported useful with strings using up to 15 delay taps spaced about 5 to 10 ms apart and spatially distributed by panning.

Doubling isn't limited to mono sources, however. A stereo pair of tracks can also be doubled, in stereo, using, obviously, a stereo delay unit. Figure 6 illustrates a

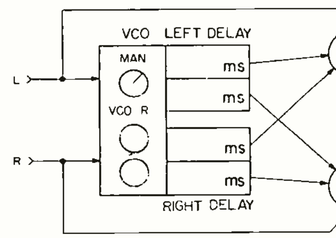


FIGURE 6
Doubling of a stereo source, into stereo, using cross delays. See text for suggested delay settings.

possible configuration for this, where the cross-crossed delays add intensity to the sound while enlarging the number of the performers and the space they appear to be in. Useful delays would be in the range of 15 to 55 ms.

Ambience Enhancement

From our discussion of room acoustics and psychoacoustics we can see how a dry mono track panned into stereo or quad lacks ambience. Delay and reverberation are the most powerful techniques available to remedy this, short of going back to live stereo or quad recording. Recall that in a live situation we hear direct sound first with interaural delay, amplitude, and spectral differences. Next, a quantity of early reflections arrive from different directions, including sides and rear; and finally the room reverberation builds up and gently decays. In contrast, the signals from close mikes are direct sound, with as little 'leakage' as can be managed, and have no early reflections or reverberation. Moreover, panned by intensity only, they all originate simultaneously with no time delay differences to provide a spatial impression.

Early Reflections

In the section on doubling, we've already seen techniques that help enhance the ambient quality of tracks at mixdown. In Figure 5, for example, if the four delays are short (say under 30 ms or so) and panned into the front, or stereo channels,

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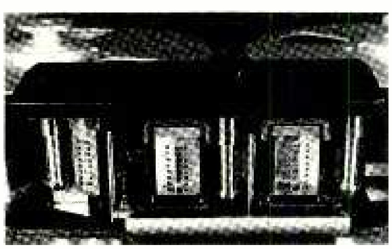

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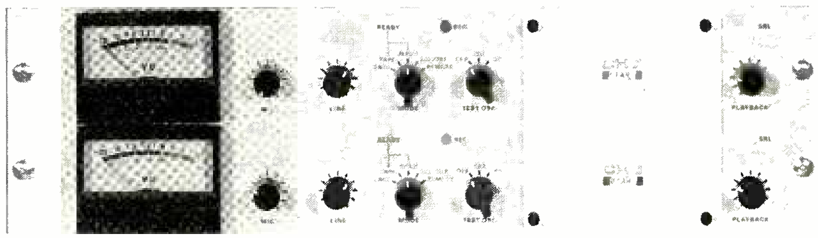

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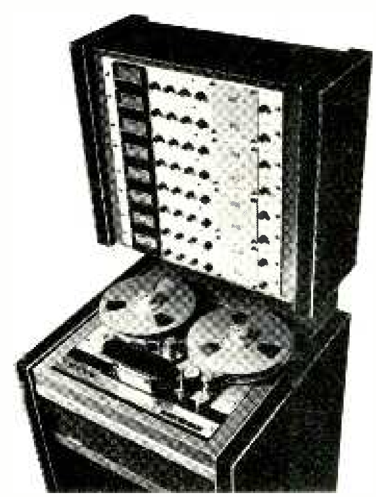

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they will simulate early reflections from the wall behind the performers, and contribute a more spacious sound to the mix. If the mixdown is to quad, the delays can be fed to the back speakers where they'll sound like side and back wall reflections and help even more. Delay times as great as 200 ms can be used for this, if intermediary delays are provided to fill in the gaps every 30 ms or so, or if the music is legato.

Delayed Echo Send

Stereo doubling can be carried into quad to establish ambience for a stereo-miked performer (Figure 7). Note the

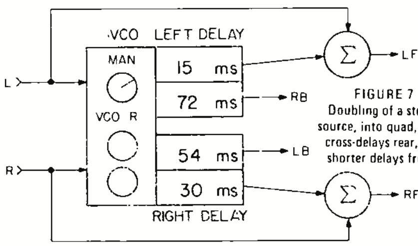


FIGURE 7
Doubling of a stereo source, into quad, using cross-delays rear, and shorter delays front.

longer crossed delays to the back channels: these help establish room size from the back-arriving reflections. Shorter delays can fill in in front, as shown, or also be panned to the back. The more channels of delay and the more taps, the farther we can go with this technique. From our room acoustics review, we learned that reverberation doesn't begin immediately, but takes time to build up. During the build-up time, a pattern of early reflections takes place, a pattern that we can mimic with digital delay taps. For reasons to be discussed later, today's digital delay units aren't versatile enough to do a good job synthesizing reverberation, but they can provide the delay needed to prevent the reverb unit from 'speaking' too soon after the direct sound. This is shown most simply in Figure 8, where we

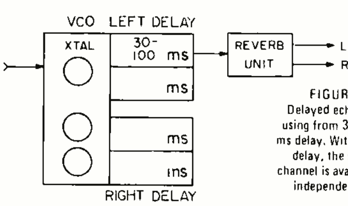


FIGURE 8
Delayed echo send, using from 30 to 100 ms delay. With a stereo delay, the second channel is available for independent use.

take the signal from one mike or some subgroup of tracks or mikes and feed it to the delay unit set for a delay of from 30 to 100 ms. The longer the delay, the greater the apparent size of the room, although of course there is a limit because the reverb will begin to break apart and arrive as an echo. With two reverb units, a stereo source, and a stereo delay unit, this can be done in full stereo.

With a digital delay unit as the delay element for the chamber we would be missing the boat if we didn't use the delay taps themselves in the final mix and thus come a little closer to simulating both the early reflections and the delayed onset of reverberation. Figure 9 shows a flow chart for a stereo pair processed with two early reflections and delayed echo. The same ideas apply to quad, where the delays could better be fed to the back channels, and where the availability of more taps would be helpful.

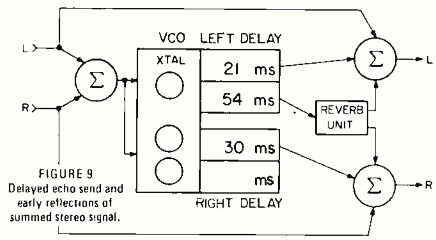


FIGURE 9
Delayed echo send and early reflections of summed stereo signal.

Figure 10 shows a stereo delay unit with four outputs connected to delay reverb and adds a delayed pair of early reflections (which could also be criss-crossed and panned back in quad).

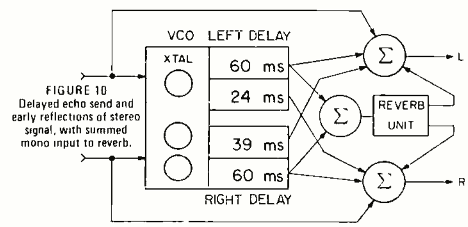


FIGURE 10
Delayed echo send and early reflections of stereo signal, with summed mono input to reverb.

Reverberation

By now there should be no confusion over the difference between delay and reverberation, although such confusion is not uncommon. A good delay unit is not an 'echo chamber' or a reverb unit: it is a device capable of providing from one to five discrete delayed repetitions of its input signal. Furthermore, a delay unit is simply not enough to serve as a basis for synthesizing high quality reverb. Special designs, utilizing high speed digital computation can produce reverberant sound, but these are not yet widely available. Nevertheless, it is natural for studio people familiar with such simple devices as the Echoplex and other tape delay-based echo units to wonder what they can do with a digital delay unit to produce something like reverberation. With a strong disclaimer to providing an alternative to EMT's and the like, we'll present some findings along these lines. Figure 11 is a flow chart for a reverb-like sound requiring a stereo delay line with VCO module.

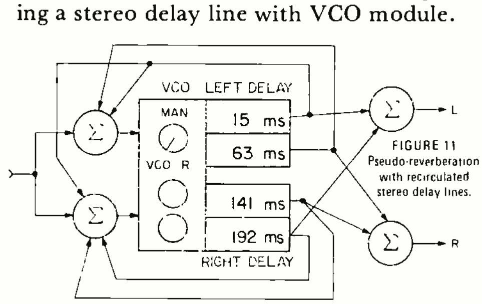


FIGURE 11
Pseudo-reverberation with recirculated stereo delay lines.

Each delay channel input is fed from a mixer receiving the same input signal. The mixer at each input also receives the attenuated signals from the delay outputs, resulting in a feedback loop with slow decay (adjustable by the amount of attenuation and delay time: short delays and attenuations produce short reverberation times). The long delay (192 ms or greater) is desirable because it allows us to achieve a given reverberation time with more attenuation. The VCO module is helpful here because it can lower the clock rate and increase the delay times while producing delay taps not related by the same increment as the fixed channel. This tends to reduce flutter echo and comb filter effects that occur when synthesizing reverb with only a few delay taps and channels as shown here. The experimenter will perhaps want to use equalization (high frequency attenuation) on the delay returns, and must adjust the relative attenuations for best effect. It may also be helpful to roll off and limit the high end of the source fed to this reverberator to prevent transients from producing a pinging or sproinging sound. Finally, add the resultant delay to the stereo mix in moderation.

Echo

Echoes occur in nature when a sound travels away from its source, bounces off a hard surface at some distance, and returns to us enough later so that we hear it as a distinct repetition. This requires a time delay of about 50 ms as we've seen, implying a distance of 27 feet. If the returning sound meets another reflective surface opposed to the first, it will again be launched away from the source and will continue to bounce back and forth until the inherent attenuation causes it to die into inaudibility. This effect can be produced from a recorded track by applying it to a digital delay unit and feeding the delay output back to a mixer at its input (Figure 12). In this way, each input signal will be followed by decaying repetitions. Adjust delay time and attenuation for the desired effect. To obtain the echo effect called 'slap' where only one or two echoes are desired, try the flow chart of Figure 13 where no feedback connection is made.

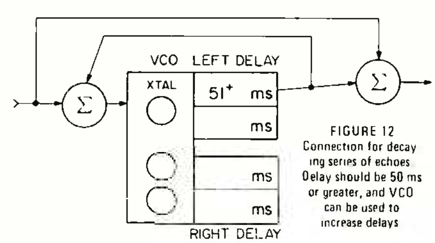


FIGURE 12
Connection for decaying series of echoes. Delay should be 50 ms or greater, and VCO can be used to increase delays

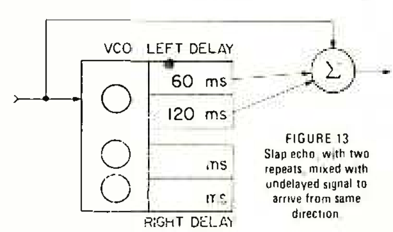
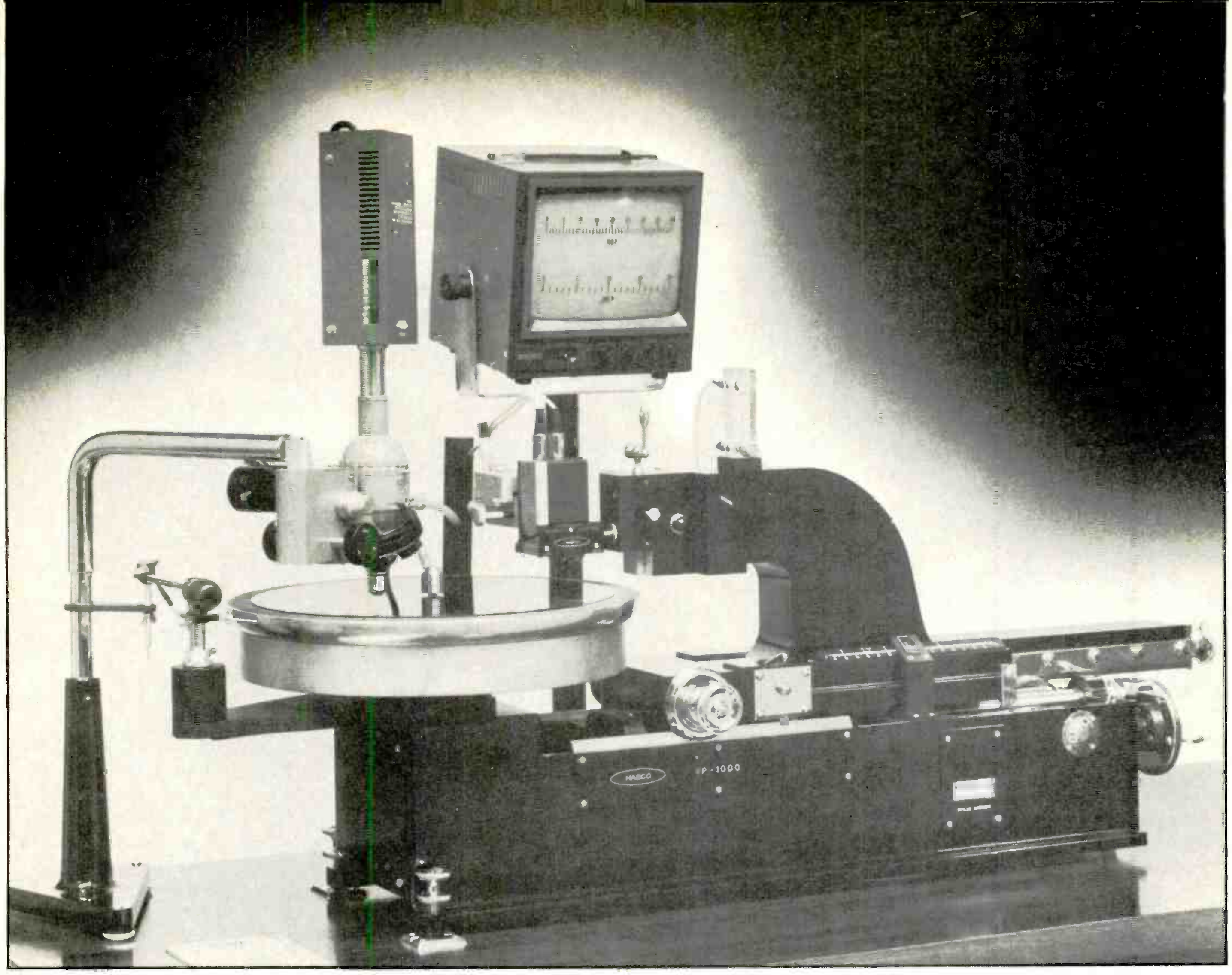


FIGURE 13
Slap echo, with two repeats, mixed with undelayed signal to arrive from same direction

In Figures 12 and 13 the echoes are shown returning to a single mixer along with the original signal, and presumably



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this composite signal will then be panned as desired in the final mix. But there's no need to limit ourselves to mono: the echoes can be panned anywhere relative to the source to produce moving echoes in space. This is especially exciting in quad where a source can be made to rotate in a circular pattern of echoes. A set-up for this is shown in Figure 14, where the source comes from LF, followed by sounds from RF, RB, and LB. A fourth delay tap comes from LF and the feedback connection takes it around the circle again. Adjust attenuation to produce the desired delay time.

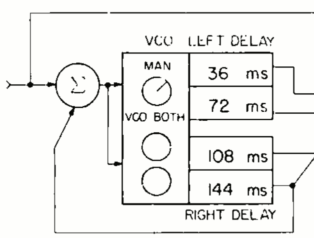

FIGURE 14
Rotating, delayed echo in quad.

A really exciting possibility exists in all these echo and slap effects to adjust the delay times so that the echoes come back in tempo with the music and fill in as faster riffs. Thus if an eighth note at fast tempo is about 100 ms long, delay taps set to 25, 50, and 75 ms can fill in thirty-second notes more accurately than any mere mortal could. Percussion instruments such as snare drums, cross sticks, wood blocks, etc., are particularly revealing of this effect. Further, the in-tempo repeats can be panned left-right or in some motion in quad for a totally unique sound. Another fertile possibility for longer delays and slap is to provide a delayed version of an existing track during overdub to the headphones of a musician who will then play counterpoint riffs off the delayed signal, to be recorded on a new track.

Flanging

Depending upon your viewpoint, flanging sounds like a jet engine, swishing, or fading multipath shortwave reception. A popular technique, flanging was first realized by recording on two tape recorders at once, simultaneously replaying the tapes and mixing the two signals. It was found that the differential delay between the two signals could be altered by placing a thumb or other suitable digits upon the flange of one of the supply reels to momentarily increase the delay.

Flanging is actually a frequency domain filtering action called a comb filter. It is so named since a comb laid on its spine with the teeth upright resembles the shape of the amplitude versus frequency curve of this filter (Figure 15). The tops of the teeth occur at frequencies where the output signal is maximum, and the bottoms at frequencies where dips or nulls occur. If the two signals are added out of phase (- sign in Figure 16), then the resultant filter has nulls at f=0, f=1/t,

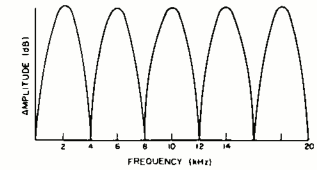

FIGURE 15: Flanging. Characteristic shape for t = 0.25 ms and negative (–) summing on connection of Figure 16. Nulls occur at f = 0, 4 kHz and at 2,3,4 . . . n times 4 kHz.

and successive nulls at 2f, 3f, 4f, etc. This is a filter with nulls for a fundamental, f, and *all* its harmonics, odd and even. Now, if the two signals are added in phase (+ sign in Figure 16), the first null occurs at f=1/2t and successive nulls occur at 3f, 5f, 7f, etc. Here, the filter removes a fundamental and only its odd harmonics. 't' in these expressions is the time delay *difference* between the fixed and variable delays. As an example, let's derive the value of t for a filter to eliminate a 1 kHz tone and all its odd harmonics. Rewriting the expression for f, we have t = 1/2f, or t = 0.5 ms, and we must use the configuration of Figure 16 with the plus sign for the fixed delay.

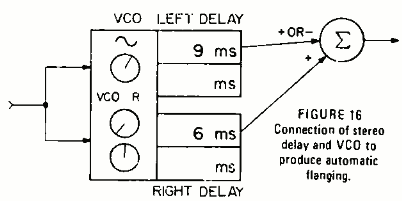

FIGURE 16
Connection of stereo delay and VCO to produce automatic flanging.

Flanging is usually done with the fixed delay summed positive, which results in more nulls. It's apparent after a bit of reflection, that longer delays produce more closely spaced nulls, beginning at lower frequencies. 50 ms, for example, produces the first null at 20 Hz and nulls from 40 Hz on up, 20 Hz apart. At first blush, you might expect that the resulting comb filter with many teeth would be most dramatic sounding, but, in fact, the shorter delays (less than 5 ms) produce the richest area of flanging sound in spite of their fewer nulls. As the null spacing increases, the nulls become broader in frequency. Changing musical pitches and spectrally broad instruments span these broad filter nulls more effectively, resulting in a quite audible effect.

Four characteristics really affect the kind of flanging sound you can get. The first is the null spacing, which should be that resulting from delays from zero to 5 ms or so. The second is that the null frequencies must be changing with time (by varying one of the delays). The third is that the musical signal must have a broad spectrum of frequencies in order to reveal the comb nulls. Flanging just isn't very effective with one voice, one flute, or even one guitar, but rather, comes into its own with synthesizers, drum kits with brushes and cymbals, and with fuller mixes consisting of combinations of instruments and voices. Finally, the flanging sound that comes from Figure 16 depends upon the gains used in the mixer. For the deepest nulls (theoretically infinite at lower frequencies), the two gains must be exactly equal so that complete cancellation occurs from the out of phase condition at null frequencies. Thus we can make flanging less obvious by reducing the level of the variable delay.

True, full fidelity flanging can be done with a stereo delay unit and the set-up of Figure 16. The right delay channel is varied from 6 to 12 ms by the VCO function, while the left channel stays fixed at 9 ms. As the VCO'd delay crosses 9 ms, sharp, dramatic flanging results. The VCO can be adjusted manually from a front panel control, programming internally from a function generator (with variable modulation frequency and range) to yield automatic hands-off flanging, programmed from an external voltage source (synthesizer), or controlled by a performer's foot pedal. For the more ambitious, the signal to be flanged or any other signal, can be externally rectified and averaged to give an envelope-dependent voltage to control the VCO and thus the flanging.

Other delay times can be used for flanging; for example, 3 ms and 6 ms, 20 ms and 30 ms, 45 ms and 50 ms, etc. The guideline is that the nominal VCO'd delay must be between one time and one-half time the fixed delay, so that the VCO whose range extends the variable delay from one to two times, can produce an overlapping delay.

Resonance

A connection for resonance is shown in Figure 17. Here, a portion of the delay output is returned back to the input, forming a recursive or feedback filter. The filter response will be similar to a comb filter, but as the return gain is increased toward unity, the comb peaks will become sharper and sharper, until unity gain, when the circuit will oscillate. This effect, which we call resonance, can be made useful and controllable if the delay element in Figure 17 is continuously variable, as it is with a VCO time base. The VCO can vary the delay either by manual control, built-in function generator modulator, or by external voltage or foot pedal. The useable control range is 2:1 of the indicated delay setting.

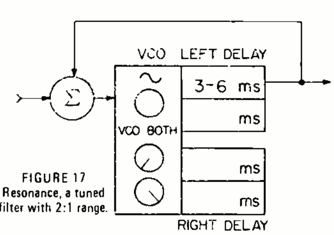

FIGURE 17
Resonance, a tuned filter with 2:1 range.

This effect is best if the source is percussive and transient in nature, such as snare or kick drum, wooden block, or cross-sticks. The resonated delay takes these broadband impulsive sounds and

The Scamp. All the signal processing you need, plugged into one box.

For example:
A compressor-limiter with push button ratio and automatic threshold selection; 30dB control range; and LED gain reduction indicator.
A sweep equaliser with three sweep sections: 20Hz to 1kHz (Q=3), 75Hz to 7.5kHz (Q=1.5), and 400Hz to 20kHz (Q=3), each with an amplitude control range of 40dB; overall LED overload indicator and output attenuator.
A parametric equaliser with the same three frequency sections as above plus variable bandwidth from half an octave to five octaves; and LED optimum modulation indicator.
A high pass dynamic noise filter with programme controlled variable slope filter (0 to 18dB/octave) with turnover frequencies at 100, 200 and 400Hz, to progressively attenuate low level, low frequency noise below the threshold; and a 20 or 40dB full frequency gating option.
A low pass dynamic noise filter similar to the high pass except it has turnover frequencies of 2, 4 and 6kHz to dynamically attenuate low level high frequency noise.
An octave equaliser providing 12dB lift or cut at the ten standard centre frequencies between 31.25Hz and 16kHz.
An LED display column with 20 LEDs per column; range of +16dBm to −40dBm; and switchable to display PPM or VU characteristics.
An expander gate with a range control to vary the maximum low level attenuation and automatically adjust the slope; a remarkably fast attack time (>5μS/10dB); a 20:1 slope gating mode; and selectable peak or rms level sensing.

The basic box is 19" by 8¾" by 13½," and is therefore standard rack mounting. It has an external power supply feeding common internal power rails. All connections are via Molex multi-way connectors.

You can build up a whole library of modules and quickly and easily transfer them from one installation to another (studio to mobile, for instance).

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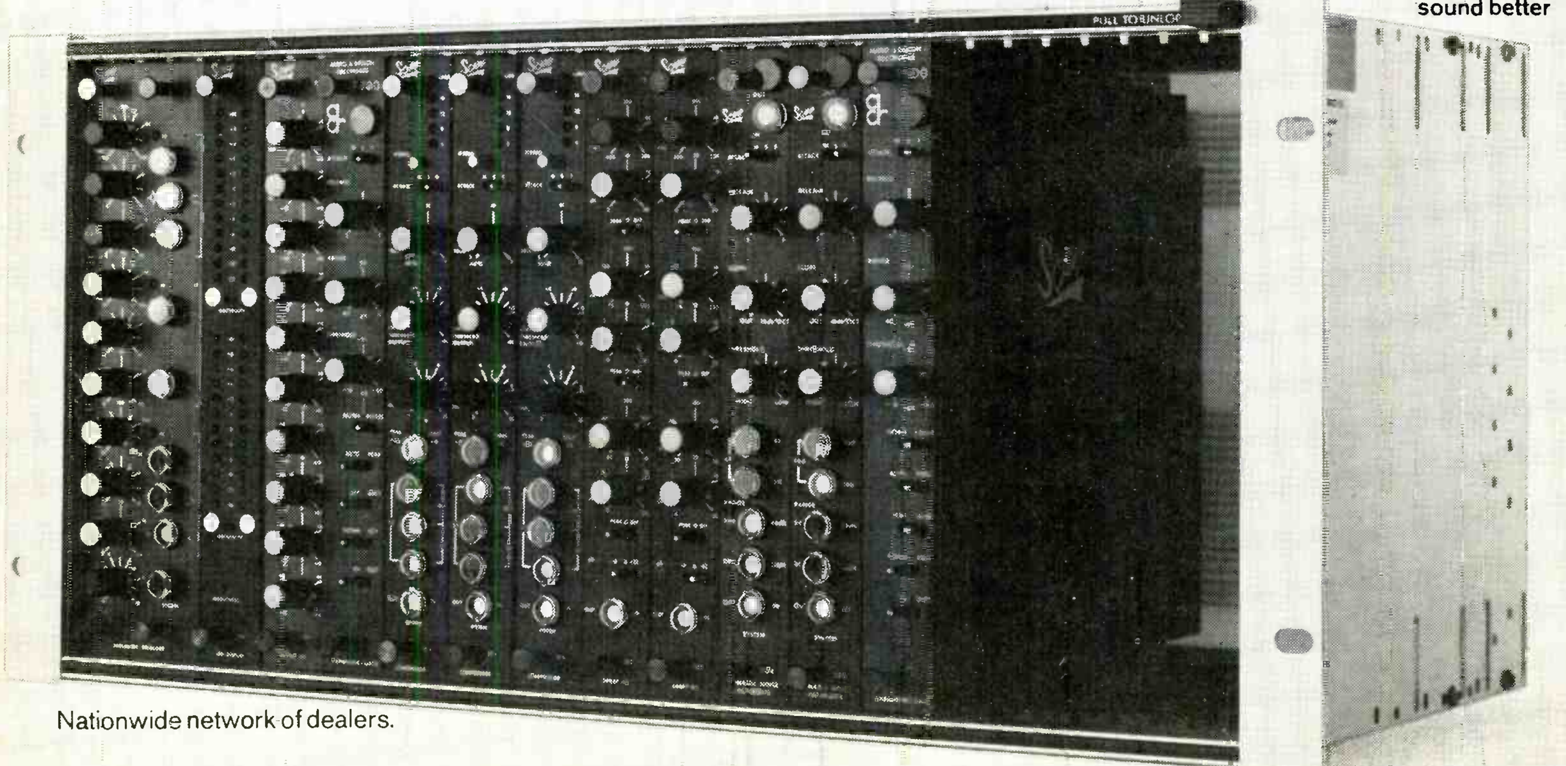


Nationwide network of dealers.

for additional information circle number 46

tunes them, due to the ringing filter action, much as a kettle drum can be tuned up and down. The difference here is that the resonant frequencies are higher and sharper and can be electronically tuned over a broad range (although only 2:1 in a single sweep). The VCO can be tuned via a front panel control, or, better yet, the percussionist can tune it with a foot pedal over a one-octave range if he has a headphone cue signal from the resonated bus.

Noise sources can also be tuned effectively by the resonance effect. Not only can a synthesizer provide the noise input to the resonant filter, but it can also voltage control the filter up and down over its octave range.

Time Base Modulation

Now that a digital delay line is available with a VCO, i.e., a voltage controlled clock, other effects are possible. We have seen already how it is useful to change the time base to a new static value, to fine tune or extend the delay time and to set up delays with a different increment. And we have used the built-in function generator low frequency modulation capability to automatically control flanging and resonance. As the rate of modulation and the amplitude (or depth) of modulation are increased, frequency modulation of the delay's output signal becomes audible. The resulting frequency modulation of inputs is easily adjusted from imperceptible to unmistakably awful, permitting the simulation of subtle vibrato to the grossest wow and flutter imaginable.

Vibrato is a slight pitch modulation at about 6 Hz usually heard on voice and violins. The modulation depth amounts to about 3% peak-to-peak for violins and about 6% for voice. A moderate delay time of around 10 ms or more is adequate to permit time delay modulation to produce the desired vibrato, while the VCO controls are set by ear. A flow chart isn't shown because the connection is simply source to delay input and one delay tap back to the mix.

With the same connection, greater peak-to-peak modulations and other modulation frequencies will produce a variety of effects. Sine or triangle modulation waveforms at low rates are recognizable as sweeps up and down, where the triangle produces fairly constant upward pitch shift during the up ramp, and downward pitch shift during the down ramp. The sine wave produces a changing pitch shift when run at very low frequencies, greatest at zero crossings. The square wave effect is interesting, in that it results in a sequence of shifted and unshifted pitches at the output. The sequence contains the original pitch, a rasied portion, and a lowered portion with no in between pitch shifts as with the two continuous waveforms.

Unless a special need for the unusual and radical transformation of a sound exists, there is probably no need to use the VCO with extreme modulation settings. But some very far out changes are possible and might come in handy. One trick is to set the modulation rate and/or delay times to coincide with the beat of the music. This allows an effect such as resonance or source pitch modulation to occur in tempo with the basic meter or shorter notes, and affect instruments only at those times. In this way, for example, the sound of a slow, metrically repeated snare or cross-stick can be subjected to resonant tuning while the rest of the instruments aren't affected.

Performer Control

We have already touched upon the possibility of giving the musician control over some time delay effects during performance. This can be accomplished with a foot pedal connected to the VCO to control the time delay over a 2:1 range. In the case of flanging, for example, characteristic spacey sound can be changed at will, faster/slower, lingering at a particular comb setting, etc. — all by the musician him(her)self actually hearing the effect on phones or speakers.

The resonance effect tunes percussive sounds. Given a foot pedal, a percussionist would be able to tune up and down over an octave range in real time. Further, even when simply using a single delay output, foot pedal effects are possible. With a basic delay setting of 20 ms, a vocalist can vary doubling over a 20 to 40 ms range, to the point where a slight echo is heard. In addition, while the musician is moving the foot pedal, a pitch shift up or down will occur that can be used to bend notes briefly. This may be of interest to performers locked into certain keyboard instruments that don't permit slurs or bending of notes. Guitarists could also make use of controllable flanging or resonance.

The VCO has a voltage input for external 0 to 5 volt sources. A synthesizer, versatile lab function generator, or homebrew circuit can drive this input and control the system clock over its 2:1 range for further performer possibilities.

Lexicon's stereo Delta-T 102 system.

The less expensive Model 92

Live Recording

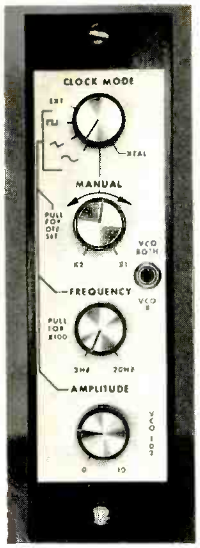
In addition to the above performer-controlled live applications, there's an interesting way of using delay during recording in a concert hall or other reverberant ambience. This one is useful for recording classical or other music live in its original acoustic setting. Recording engineers have for years noted the conflict between the close pickup needed to highlight or balance certain instruments and the more remote pickup needed to capture the overall orchestra embedded in room sound.

The conflict arises because the accent mikes, which pick up the instrument's sound with only about one millisecond of time delay, are mixed with the more distant mikes which also pick up the same instrument with perhaps 30 to 50 ms of time delay. Thus the mix can contain a faint doubled version or echo of the highlighted instruments or vocalists. An interesting solution is to use a delay unit for the more important accent signals so that it feeds the mix in near synchronism with the distant pickup, or perhaps a few ms earlier. This technique has been known for some time, but has not become practical until the advent of delays high enough in quality to permit their use with the wide dynamic range signal from a good microphone. A stereo unit is particularly attractive for this job because of the economical packaging of two independent channels, each with a modest amount of delay and only one output.

Selecting a Delay Unit

There are a number of delay units available at present, and more on the way. Some important considerations in choosing a delay unit are: audio performance, features, and flexibility. The Lexicon Delta-T 102-S is one unit with an excellent combination of features and performance for studio applications.

Audio performance, or signal fidelity, involves several factors, including frequency response, noise, distortion, and dynamic range. Today, the best delay lines are digital; that is, the audio signal is first time-sampled and then quantized. The process of digitizing must be done extremely well, as the noise and distortion resulting from quantizing can be more annoying than a numerically equivalent amount of analog noise. Virtually all contemporary delay lines (digital as well as analog) use companding or floating point techniques to obtain a wide enough dynamic range. The choice of companding method, the inherent noisiness of the delay line, and the overall quality of the engineering design determine how quiet the result will be. Lexicon uses an exclusive, proprietary, floating point digital encoding that results in a 90 dB dynamic range and extremely low distortion over a wide range of signal levels. Not only does the 102-S specify a 1 kHz maximum level distortion and noise of less than 0.2%,



Close-up of Lexicon's VCO 102 detailing various front panel controls. The top control chooses Xtal for normal operation with fixed time base; manual for control of delay over 1X and 2X range by knob below; three waveforms for modulation by built-in function generator; and ext. for external voltage or footpedal control of delay over 1X to 2X range. Frequency and amplitude together with the waveform, control the function generator that modulates the VCO, permitting modulation from .2 Hz to 2 kHz in frequency and from zero to 1X-2X delay peak-to-peak. The toggle switch selects both channels or right channel only for VCO control.

but it also achieves distortion of less than 0.3% 34 dB down from maximum level. This ensures that even quiet passages are not degraded by quantizing noise. Other manufacturers fall short of the high level distortion and noise spec, and conveniently omit the lower level performance. In addition, the floating point encoding guarantees that the characterisitically gritty and signal-dependent noise at levels near the bottom end of the digital range occurs 90 dB down and is thus inaudible. Thus it can be seen that the dynamic range and total distortion and noise specifications are worthy of very careful attention for those desiring a high level of audio quality in a digital delay.

Good frequency response in digital delay units calls for conservatively high sampling rates coupled with accurate, multipole, low pass filters at input and output. The Delta-T 102-S uses Butterworth, maximally flat filter designs to yield smooth, flat response out to the edge of the passband, rolling off with a good margin well before the Nyquist half-sampling frequency to prevent aliasing during input sampling. Good attention to design is essential to keep spurious digital clock frequencies out of the output signal, and requires not only the already mentioned filters, but excellent layout and grounding. In analog delay designs, e.g., those using bucket brigade devices, it is extremely difficult to keep a smooth, extended frequency response as the clock is varied to change delay. In addition, the noise increases steadily as a function of the product of delay time and frequency response. A digital design such as the 102-S reduces these problems to non-audibility at A-D conversion, after which they are independent of the delay time throughout the unit's frequency range.

When it comes to features, the ultimate test is how much can be done with a given delay unit. Again the 102-S provides an example of useful versatility. The two delay lines can be clocked together with crystal and/or VCO for special effects. As can be seen from this article, the combination of a VCO with a two-channel system is extremely powerful.

Modular construction offers the user a unique combination of economic value and an in-built hedge against obsolescence. At time of purchase, a studio with a limited budget can elect to buy only one channel if desired, and expand into full stereo operation later, with simple field installation of modules. Since the delay or memory capability is modular, it, too, can be chosen frugally initially and expanded in 48 ms bites later on.

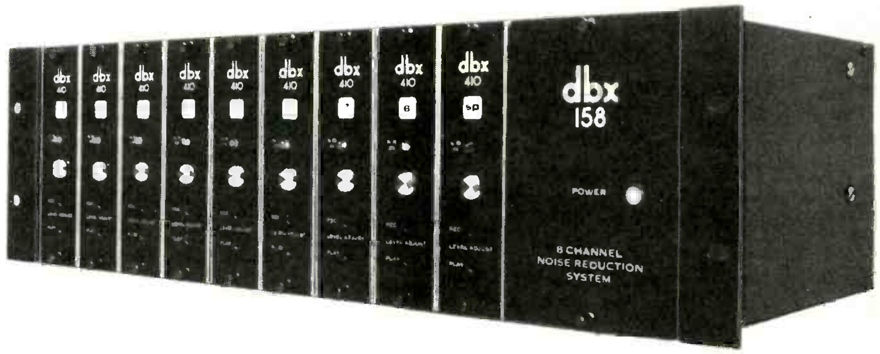
Putting it all together, performance, versatility, and reliability, coupled with cost and modularity (i.e., buying only what you need today with the option for economical future expansion) are the key factors upon which a selection will be made. The Lexicon Delta-T 102-S is a good exmaple of equipment that embodies these key factors.

Note: R-e/p readers can receive a complimentary, complete application note with a companion demonstration record by writing LEXICON on their letterhead, requesting the AN-3 Application note with demo record. The record contains several of the effects described in the article. Write:

LEXICON
60 TURNER ST.
WALTHAM, MA. 02154

New Products

dbx ECONOMICAL EIGHT CHANNEL TAPE NOISE REDUCTION SYSTEM



The new dbx 158 provides eight channels of simultaneous record/play tape noise reduction in a modular rack mount package at an economical price. The system offers the semi-professional recordist or small studio all the advantages of dbx professional systems, including 30 dB of noise reduction and 10 dB of additional recorder headroom, and is a classic 2:1 mirror image compander which compresses the signal's dynamic range of the tape recorder. At playback the system fully recreates the original input signal with the complete lack of audible tape hiss usually encountered in the recording process.

Model 158 is a modular design with one channel of independent record/playback electronics in each module. This feature permits simultaneous monitoring of the noise reduced program while recording is in progress. A free spare plug-in module is supplied in the main frame to eliminate down-time.

This system accepts the outputs of any preamp or mixer up to 50 K ohms and drives the line level inputs of any Tascam, Otari, Brenell or other multitrack recorder having single ended inputs and outputs terminated in RCA type phono connectors. No level match tones or routine adjustments are required for accurate record/playback operation.

Model 158 is fully compatible with all dbx professional studio tape noise reduction systems and tapes made on the 158 may be played back on dbx professional, semi-professional and on-board TEAC dbx units with complete interchangeability. The system occupies only 5¼" of rack space and its light weight of 17 pounds makes it convenient for use on location recording jobs.

The system, including the spare plug-in module sells for $2,400 and is available from dbx professional audio dealers. For complete information and list of demonstrating dbx dealers, circle reader service number or contact:

dbx, INCORPORATED
71 CHAPEL STREET
NEWTON, MA 02195

for additional information circle number 49

IE-15A DISTORTION ANALYZER ADDED TO IVIE AUDIO ANALYSIS SYSTEM

The IE-15A Distortion Analyzer is an accessory designed to be used with the IE-10A Audio Spectrum Analyzer. The IE-15A now adds total harmonic distortion (THD) measurement capabilities to the already powerful IE-10A. In combination you can measure THD at a fixed frequency of 1 kHz over the range of 100% to less than .02% with an accuracy of 1.0 dB. A clip-on IE-10A overlay screen is provided that allows the operator to read the distortion levels directly in % or dB.

Built into the IE-15A is an ultra-pure sine wave source having selectable frequencies of either 1 kHz or 4 kHz and output voltages adjustable over a 60 dB range. In addition to distortion testing, the oscillators can be used in troubleshooting applications, or in any other task requiring a stable, clean signal source.

for additional information circle number 48

Professional perfection in most exacting microphone applications

Everyone enjoys a contest... particularly when they've got a winner. We are ready to take on *anybody* with our latest AKG large diaphragm professional condenser microphone—the C-414EB.

With many exclusive features, it will proudly outperform its nearest rival, yet costs *two hundred dollars less!* The C-414EB has four selectable polar patterns; improved maximum sound-pressure level capability through a built-in 0, -10 and -20dB attenuation selector (greater than 155dB sound pressure level capability); a three-position low-frequency rolloff switch with 14dB/octave slopes; freedom from off-axis coloration and uncompromisingly smooth and natural sound characteristics—plus a new level of robustness and dependability.

We've given it the acid test. Dust off your alchemy set... you're in for a pleasant surprise!

C-414EB microphone shown above with optional H-17 shock-mount/windscreen

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PHILIPS AUDIO VIDEO SYSTEMS CORP.
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C-414EB HIGHLIGHTS

- Attractively priced
- 4 switchable polar patterns—
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  - omnidirectional
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  - hypercardioid
- 3-position attenuator (between capsule and preamplifier) 0, -10dB, -20dB
- 3-position bass rolloff switch, flat, 75 Hz, 150 Hz (-3dB points)
- 12V/48V phantom powering
- H-17 shock mount/windscreen assembly for superior isolation from low-frequency vibration and wind noise or pops.
- Standard three-pin XLR type connector
- Fully RF shielded
- Extremely quiet (equivalent noise level: 20dB SPL)

Full specifications are available on this and all AKG microphones on request.

for additional information circle number 50

The IE-15A oscillators have a separate on/off switch for the measurement of signal-to-noise ratios over a dynamic range of 100 dB.

Other IE-15A features include an overload indicator light to prevent measurement errors, and an output monitor jack for viewing distortion products on an oscilloscope.

Rechargeable nickel cadmium batteries power the IE-15A for 12 hours between charges and the AC adaptor/charger provides continuous line operation from either 115 or 230 VAC.

IVIE ELECTRONICS INCORPORATED
500 WEST 1200 SOUTH
OREM, UTAH 84057
PHONE: (801) 224-1800

for additional information circle number 51

FLAT FREQUENCY RESPONSE AND NEW SHOCK MOUNT SYSTEM ARE FEATURES OF NEW SHURE MICROPHONE

Called the SM59, the microphone is a dynamic type with a wide 50-15,000 Hz frequency reponse that provides clean, natural reproduction without a presence peak in the higher frequency range. This feature, coupled with its anti-feedback, cardioid pickup pattern makes the SM59 perfect for use in studios, live performances, churches, and meeting rooms.

Another major feature of the SM59

is its patented mechanopneumatic shock mount system that dramatically reduces mechanical noise and pickup of floor and desk stand vibrations. A special "pop" filter also protects against explosive breath sounds.

In addition to its outstanding performance characteristics, the SM59 has a slim, sleek appearance. It weighs only 215 grams (7.6 oz.), is just 197.3 mm (7-25/32 in.) long, and is ruggedly built to withstand rough use both indoors and out.

User net price of the SM59 is $132.00.

SHURE BROTHERS, INC.
222 HARTREY AVENUE
EVANSTON, IL 60204

for additional information circle number 52

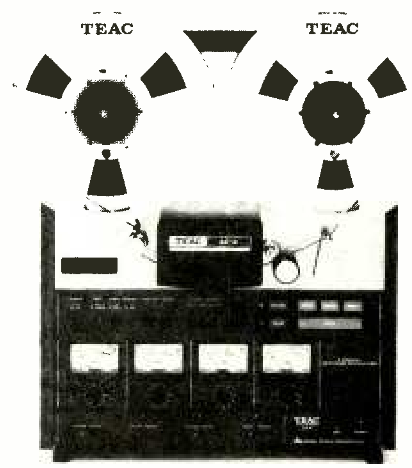
for additional information circle number 53

TEAC TASCAM SERIES MODEL 40-4 1/4 INCH FOUR-TRACK RECORDER/REPRODUCER

The newly introduced rugged ¼ inch four-track unit features the same transport and chassis as the eight-track 80-8 unit.

Additionally, the three-head 40-4 features full integrated circuit logic with motion sensing and a memory stop function.

According to Bill Cawlfield, TEAC's director of product dvelopment, the new unit has a nationally advertised value of less than $1,600 and will be available in quantity for national distribution in August.



Cawlfield said the 40-4 has a combination record/reproduce head, and erase and monitor heads, function and output select buttons, LED overload indicators, accessible calibration controls and a flip-up head cover. The unit has optional four-channel dbx (DX-4), remote control (RC-170) and mike preamp module (MA-4).

The unit takes up to 10½ reels and records at 15 and 7½ ips. It has dual-speed, hysteresis synchronous capstan motor and two eddy current induction reel motors. Wow and flutter is 0.05% (NAB WTD) at 15 ips and 0.07% at 7½ ips, frequency response of 50 to 20,000 Hz at 15 ips and 50 to 15,000 Hz at 7½ ips, a signal-to-noise ratio of 65 dB weighted, and distortion of 1% at 1,000 Hz.

TEAC CORPORATION OF AMERCA
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
PHONE: (213) 726-0303

for additional information circle number 54

TROUPER I LIVE MUSIC MIXING SYSTEM

Like its bigger brothers, the Trouper I is a modular system. The Output Control Module ($749 suggested retail value) has eight inputs and the Expander Module has ten for $698 (suggested retail value). The Trouper I is said to be accessible for just about anyone who is interested in putting premium sound in their performance. It's built to handle the durability of the road as well as having the specs that you expect from a Uni-Sync product.

Home Cookin'!

See that guy at the board? Once upon a time he was an engineer at the busiest studio in town. The place had everything big money could buy. And it cranked out super-slick albums at an absolutely psychopathic rate. But because its hourly rate matched its image, it wasn't only the busiest studio in town, it was also the most expensive. Which was alright if you had a fortune to spend—which the band you see here didn't.

After years of being a staff engineer he decided he'd been sitting behind somebody else's board long enough, thanks. So with the money he'd saved, he invested in a complete TASCAM Series recording studio by Teac—80-8* eight track, 25-2* two track, mixing consoles—the works!

Two days later, he was making tracks like they'd never been made before—in his home! And at a fraction of the price charged by his former employer.

Well, the band cut a demo at his new studio and with it they got a record deal. And with the front money, they invested in their own Teac mini-studio. So with the band members taking turns at the board, they laid down the tracks for their album. And to make sure they got the most out of the tracks they made, they asked the old pro to do the final mix.

Could this story have happened without Teac recording equipment? Not on your life. But it's the sort of thing that's bound to happen whenever a second generation engineer and a second generation band team up with a new generation of recording instruments.

TASCAM SERIES BY TEAC

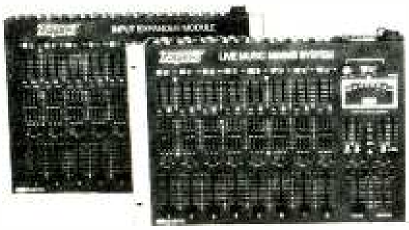
A new generation of recording instruments for a new generation of recording artists.

*Nationally advertised value. Model 80-8 tape recorder shown above, less than $3,000. Model 25-2 tape recorder also shown, less than $1900 (Rolling Consoles not included). Actual retail prices to be determined individually at the sole discretion of authorized Teac Tascam series dealers. Prices subject to dealer preparation charges where applicable.

TEAC Corporation of America, 7735 Telegraph Rd., Montebello, CA 90640 ©TEAC 1977

for additional information circle number 55

Each input contains the following features which separate the Trouper I from other mixing systems: low level balanced input, High Z Input, In/Out Jack, 20 dB of mike attenuation, monitor send, echo send, three band graphic equalizers, solo switch for individual channel monitoring by the operator (incidentally the VU meter follows the solo system giving you a visual as well as audible indication of each input and output), and the input mix control.



The Output Control Module besides having eight inputs with the same features as above include, House and Monitor Outputs, Echo Send/Receive Jacks, Headphone jack, Solid State LED VU Meter, Echo Send Master to the built-in reverb, Headphone level control, House/Monitor echo receive, High/Low Cut Filters, and House/Monitor Level Controls.

The Trouper I Expander and Control Module couple together simply by console interconnection and the supplied umbilical cable. It gives you 18 inputs of quiet, heavy duty mixing ability. It is rack mountable and available in heavy duty carrying cases built for the road, choose between two models holding one unit each or holding both expander and output control module.

UNI-SYNC, INC.
742 HAMPSHIRE ROAD
WESTLAKE VILLAGE, CA 91361
PHONE: (805) 497-0766

for additional information circle number 56

NEW ASHLY LIMITER FEATURES ULTRA-LOW NOISE AND DISTORTION

Designed to fit a variety of applications, the new low-priced model SC-50 peak limiter/compressor features accurate, independent adjustment of all AGC characteristics. This is accomplished using a specially designed closed loop detector circuit which keeps the output ceiling accurate at high compression ratios, yet remains smooth down to a gentle 2:1 ratio. The wide range of attack, release, and ratio adjustment allows tailoring of the limiting action to suit any program source. A program dependent dual release time action provides quick recovery from isolated transients while allowing slower release from sustained overdrive; this eliminates audible "pumping" and "breathing" effects associated with many

peak limiters. Also featured are ultra-low noise (-90 dBV), and distortion (<.05%).

The SC-50 can be left in the program line as a safety device with virtually no change in signals below the limiting threshold.

A unique LED display indicates gain reduction and threshold. Two or more limiters may be connected to provide accurate tracking in stereo or quad applications.

Suggested applications include loudspeaker protection, vocal compression, broadcast limiting, loudness enhancement, and a variety of special effects such as musical instrument sustain.

Suggested list price for the SC-50 is $299.00, F.O.B. Rochester, New York.

For additional information, please contact:

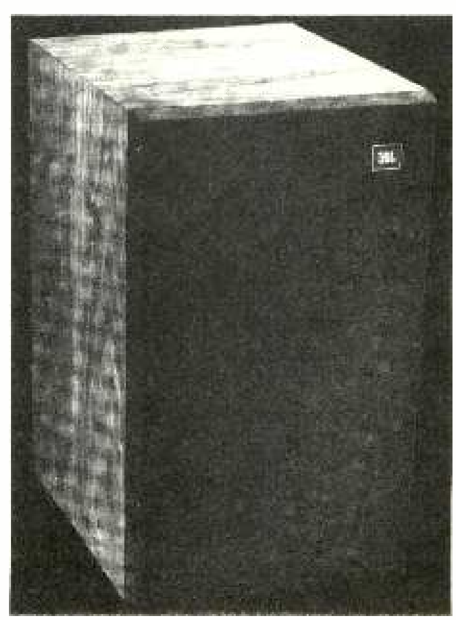
ASHLY AUDIO, INC.
1099 JAY STREET
ROCHESTER, NY 14611
PHONE: (716) 328-9560

for additional information circle number 58

JBL UNVEILS BROADCAST MONITOR LOUDSPEAKER SYSTEM

The 4301 delivers wideband sound reproduction, in addition to high efficiency and accuracy. Use of the 4301 in broadcast studios is particularly relevant in light of the most recent developments, including television/FM stereo simulcasting AM stereo and multiplex television audio.

Its 8-inch (20 cm) low frequency loudspeaker was engineered specifically for use in compact enclosure without the compromises usually associated with smaller drivers. For its size, this unit exhibits unusually smooth frequency response, wide dynamic range, superior transient reproduction and low distortion. It features a precision die-cast aluminum



for additional information circle number 57

frame for structural integrity under the most severe operating conditions. Its 2-inch (5 cm) voice coil is suspended in a magnetic field having a flux density of 0.85 tesla (8500 gauss).

The magnetic field is generated by a 2½ pound (1.1 kg) low loss magnetic assembly energized by an Alnico V magnet. Mass and compliance of the integrally stiffened cone have been carefully selected to optimize low frequency bandwidth and definition while reducing distortion.

High frequencies above 2,500 Hz are reproduced by a 1.4-inch (3.6 cm) direct radiator designed for clarity, smoothness of response and power handling capacity. Its 5/8-inch (1.6 cm) copper voice coil is large in relation to cone size for efficiency and accurate transient reproduction, yet the diameter of the cone and center dome is small enough to maintain wide dispersion at extreme high frequencies. The voice coil is suspended within a 1-5/8 pound magnetic assembly that generates a flux density of 1.5 tesla (15,000 gauss).

The system has a nominal impedance of 8 ohms and has a rated power handling capacity of 15 watts continuous sine wave.

The 4301 enclosure utilizes a ducted port for proper acoustical loading of the low frequency loudspeaker. Its dimensions are 19" x 12-1/6" x 11-1/4" deep, making it suitable for horizontal mounting on a standard rack shelf. The enclosure is finished on four sides in hand-rubbed American Black Walnut veneer complemented by a dark blue fabric grill.

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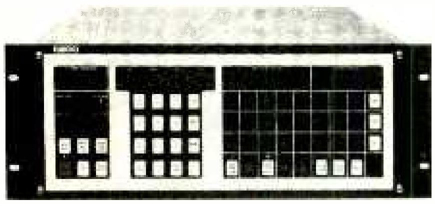
for additional information circle number 59

NEW EECO AUDIO/VIDEO TAPE SYNCHRONIZER

EECO's new, microprocessor-based MQS-100 series synchronizing system can cue and synchronize any three mag tape transports including video, audio and mag film simultaneously. The SMPTE/EBU Edit Code, used for indexing of the tapes, need not be identical and tapes with drop-frame and non-drop-frame formats can be intermixed.



System modes include High Speed Search and Cue, follow the leader or "Chase Mode", Synchronized Play Back, Fast and Slow Re-synchronization and Roll-Back with automatic re-synchronization. Operational efficiency is demonstated by control simplicity. One button actuates all transports to roll back, start forward and synchronize automatically.

Time code readings for all tapes can be "captured on the fly", individually or simultaneously. A plus or minus offset of any selected time increment can be preset for each slave transport.

The "Chase Feature" of the MQS directs the slave transports to follow all master transport actions. This permits the operator to control cueing and synchronizing at the front panel of the master transport.

The MQS is 7" high with standard 19" wide Retma Mounting. For more information contact your local Ampex Sales Office or:

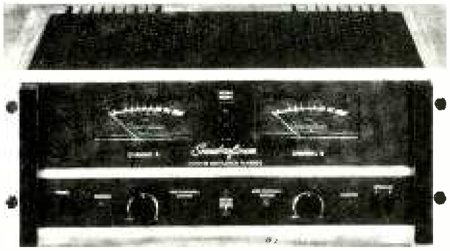
EECO
1441 EAST CHESTNUT
SANTA ANA, CA 92701
PHONE: (714) 835-6000

for additional information circle number 61

NEW CLASS "H" AMPLIFIER ANNOUNCEMENT BY SOUNDCRAFTSMEN

A "New Class" Super-Power Amp (250 watts RMS per channel 20-20 kHz both channels driven into 8 ohms, less than 0.1% THD) was introduced at the May Audio Engineering Society Show using a revolutionary new Patent Pending "Vari-Portional" system of Analog Logic Circuitry which anticipates power demands and supplies only a proportional amount of power, as required by varying input signal voltages.

Invented by Soundcraftsmen's Chief Engineer, Paul Rolfes, holder of seven patents in the field of solid state electronic power circuitry, together with his assistant, John Holyoake, the advantages of the "Vari-Portional" system are obtained through it's constant metering of output power requirements for optimum efficiency. This results in direct and measureable energy savings by reducing the amout



of energy dissipated as heat loss, yet with controlled full power always available, standing by, and supplied as needed. This higher powered amp can be sold at a price even lower than ordinary class AB amplifiers through cost savings made possible by the Patent Pending "Vari-Portional" circuitry. For example, no fan is needed even under most severe operating conditions.

An added advantage is a substantial savings in power consumption. Class AB amplifiers of the same power rating, operating at 1/3 power in accordance with FTC test requirements, will consume over 40% more energy than the Soundcraftsmen "New Class" amplifier. Thus, the "New Class" amp provides savings in heat dissiaption of approximately 200 watts. Progressively greater percentages of savings may be obtained at lower power levels.

Other outstanding performance features of the Soundcraftsmen "New Class" amplifier are its uniquely designed exclusive all solid-state "Crowbar" fail-safe overload protection circuitry with automatic reset, (no circuit breakers or fuses), for 100% protection in the event of shorted speaker leads, etc. Totally non-limiting output circuitry eliminates any possibility of limiter-caused distortion due to excessive current demands.

Performance specifications are conservatively rated at less than 0.1% THD, noise better than 105 dB down, and input sensitivity for full output, only 1.28 volts. Slew rate better than 25, damping factor greater than 100, frequency response 0.25 dB, 20 Hx - 20 kHz. Power rating 250 watts RMS per channel into 8 ohms 20 Hz - 20 kHz both channels driven. Suggested retail price is $699.00.

Because Soundcraftsmen has recognized the need for a basic, rugged, indestructible super power amp in professional applications, a "no-frills" model is also available. Model PA5001, with the same new class circuitry and specifications but without many of the front panel convenience features, has a suggested retail price of only $549.00.

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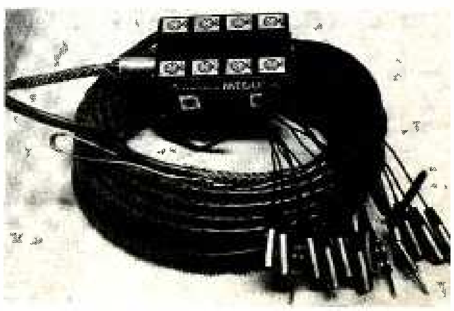
1721 NEWPORT CIRCLE
SANTA ANA, CA 92705
PHONE: (714) 556-6191

for additional information circle number 62

MEDUSA FOR MULTIPLE WIRING

Medusa, from Whirlwind Music, simplifies the multiple wiring needs of PA and recording systems setups, according to the manufacturer. Fabricated of the highest quality materials and designed by professionals, Medusa is available in eight standard setups, ranging from 50-foot cable, six mikes in with three sends, to 100-foot cable, 24 mikes in, with three sends.

The manufacturers say they will custom design and make virtually any Medusa to fit the customer's special needs. Medusa stage boxes are also available in a wide selection of basic sizes and con-



for additional information circle number 63

nector configurations. Standard inputs are XLR female, but others may be specified. Phone jacks can be specified in ¼-inch phone jacks or XLR male configurations.

Other options include impedance matching and splitter/transformers.

WHIRLWIND MUSIC
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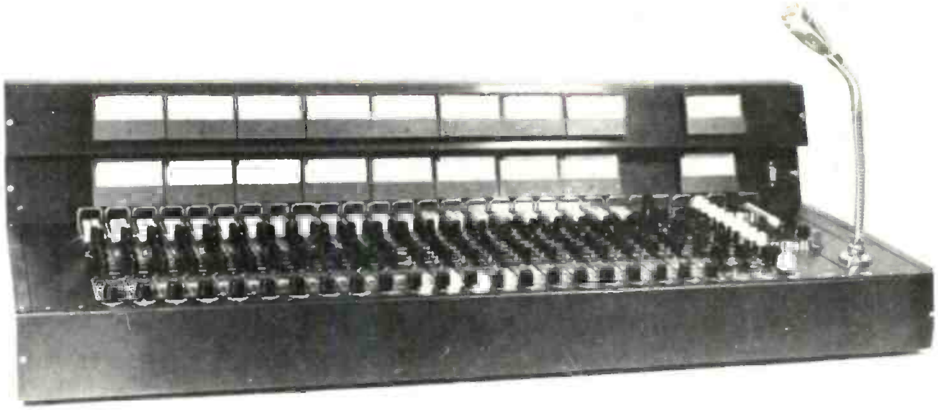

INTERFACE HIGH PERFORMANCE LOW COST 16 TRACK MIXER

Interface Electronics of Houston, Texas, has announced the Series 316 sixteen track recording consoles: available in 24 or 32 input mainframes, these high performance low cost mixers provide pushbutton selection of 16 track outputs plus four independent cue/echo sends; input modules provide flexible equalizing including (in the 316B) a parametric midfrequency equalizer tuneable from 150 to 7,500 Hz. Also included are phantom power, phase reverse, panpot, six step preamplifier gain set switch with two input pad positions, and long-travel Duncan conductive plastic slider attenuator. Console normally includes mixdown for control room monitor with solo to monitor only, also track masters and talkback/slate module with gooseneck microphone. Output section options include ouput transformers and dual octave graphic equalizer. Mainframe includes 18 four-inch lighted VU meters having industry standard response, ultra reliable regulated power supply, and masters. Construction is entirely modular and plug-in and uses new high-slew rate internally compensated plug-in integrated circuits throughout. Numerous options are available in both input and output sections, and mainframes may be ordered with less than a full complement of input modules and more modules added later.

Performance is equal to or better than consoles costing many times more, yet circuits are kept simple in order to improve reliability. Thorough shielding and balanced input transformers assure against pickup of radio transmissions or noise from SCR light dimmers. Overall response is within ±1 dB from 20 to 20,000 Hz. with ±12 dB of equalizing available at specified frequencies. 400 Hz. distortion is under 0.1% THD up to almost the clip level, and maximum level is approximately 19 volts RMS with bridging load; since zero level is 1 volt RMS, the headroom is of the order of 20 dB. Equivalent input noise is approximately 0.5 microvolts, or -126 dBm. Prices of complete mixers start at $8,870.00, and delivery is currently eight weeks.

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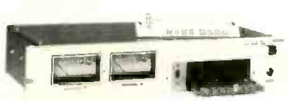

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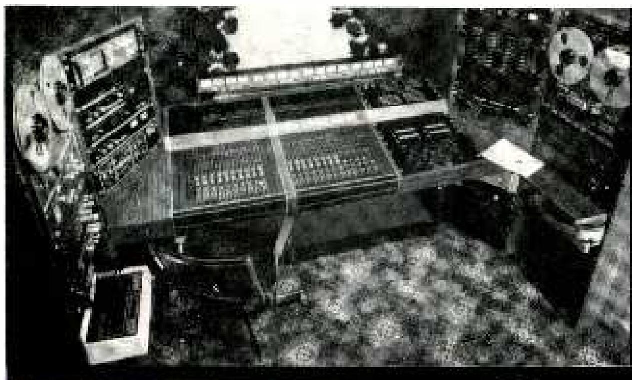

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– BOOK REVIEW –

SOUND RECORDING
by John Eargle

Sound Recording by John Eargle touches on nearly every aspect of contemporary audio recording of general interest. It is aimed at the modern recording engineer/producer, who is often short of technical background, though perhaps long in musical production experience.

The book introduces the reader to basic acoustic and recording principles, and gives an historical background to the development of the recording arts. Basic recording tools are described as well as the newer developments the tools have made possible. The language is relatively nonmathematical, but definitely the language of a knowledgeable and insightful engineer.

A chapter on the physics of sound moves quickly and easily through basic principles, especially those closely associated with recording environments. The subject of psychoacoustics is introduced, which leads into a discussion of stereo and quadraphonic sound, and the various associated systems.

Recording equipment is covered in the last six chapters. Microphones are dealt with in depth, with a particularly interesting treatment of directional patterns. Monitor systems and environments are described with special emphasis on modern recording requirements. A chapter on control systems explains the function of the recording console, and deals with topics like gain structure, noise, reference levels, metering, and automation.

The principles of magnetic recording are briefly explained, and the simpler aspects of modern recording machines are described. A chapter on signal processors covers the how's, why's and when's of filters and equalizers, compressors, limiters, and expanders. Noise reduction is discussed with attention to the relative advantages and disadvantages of the various systems. Reverberation and digital time delays are explained, and there is a brief treatment of some special effect devices such as ring modulators, etc.

Disc recording and reproduction winds up the book, touching briefly on the many aspects of the art and technology of cutting.

There is no mention of digital recording techniques and equipment, nor any reference to the recording equipment, both optical and magnetic, used in the making of motion pictures. In fact, there is little specific information about multitrack recording at all. But there is more information in the book than one realizes in reading once through. Mr. Eargle's long experience in the industry shines through in his ability to relate the principles and equipment of recording to the act of making recordings, and that's what engineering is about, isn't it?


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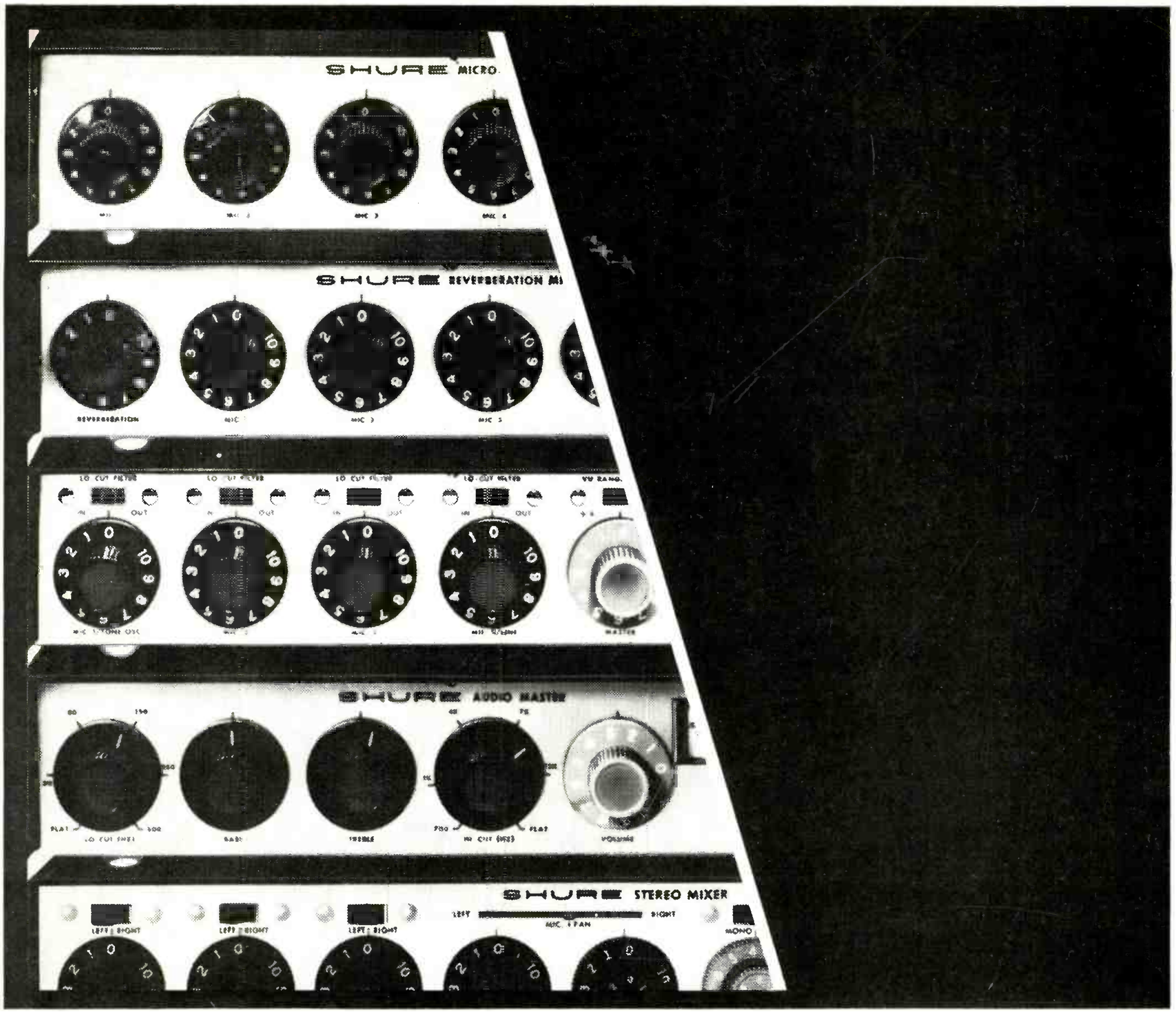

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