AUDIO PRODUCTION FOR BROADCAST

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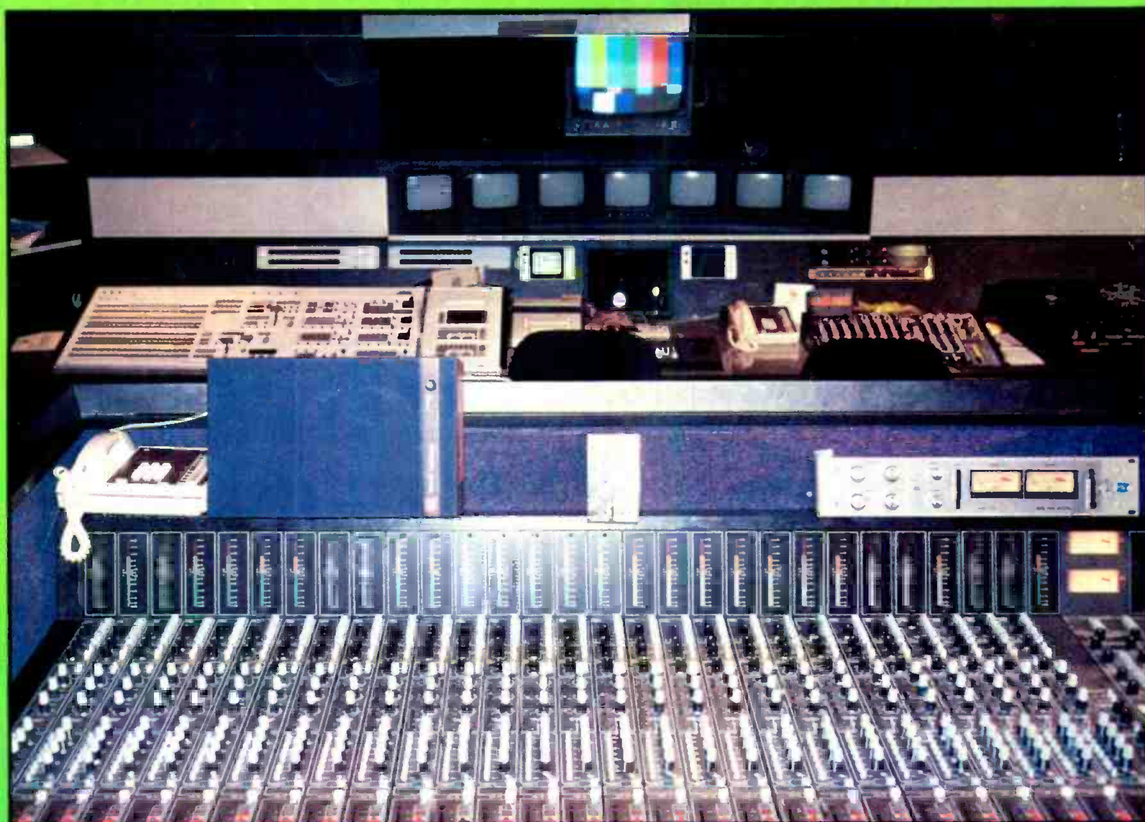

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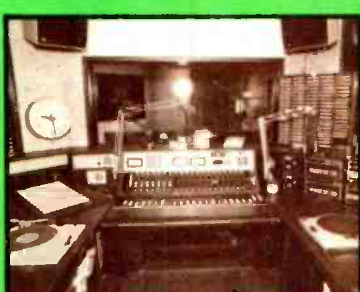

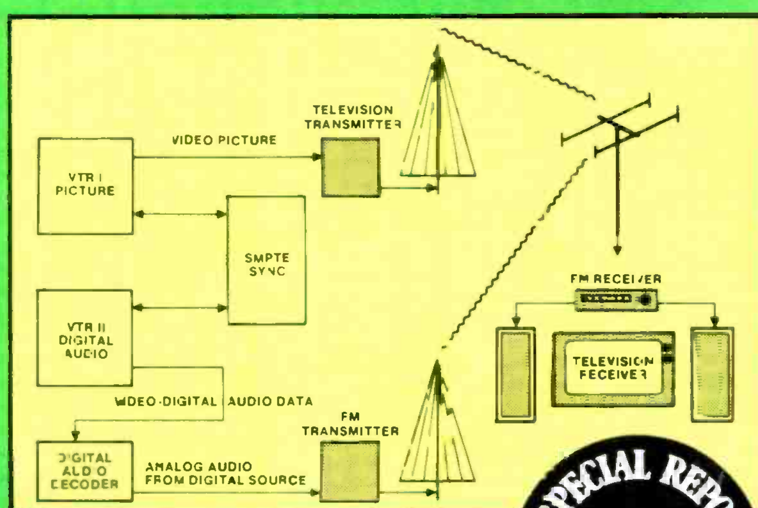

A SPECIAL REPORT FROM THE EDITORS OF RECORDING ENGINEER/PRODUCER

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— Contents —

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MULTITRACK AUDIO PRODUCTION STUDIOS

Four- and Eight-Track Experience on the West Coast

by Douglas B. Howland, Consulting Editor

Multitrack production is a relatively new venture for broadcasters, when compared to its widespread and routine application throughout the recording industry. The stations interviewed for this article have used multitrack techniques only in the last 10 years, and their production formats vary from two- to eight-track. While a certain number of broadcast production studios around the country do make use of more than eight tape tracks, they are still rather few in number, and the majority utilize four-track machines.

Radio's product is obviously the material broadcast over the air, which may include live and/or pre-recorded programs, commercials, public service announcements, promotions, and so on. Multitrack production techniques have enabled broadcasters to produce a higher quality product — both technically and creatively — for the advertiser that supports the station's income, and for the listener who in turn supports the advertiser.

Creativity is enhanced when station staff have the facilities to undertake complex productions. Multitrack can eliminate the time-consuming process of synchronizing many individual machines, and performing multiple sound-on-sound dubs. Individual elements of the final production can be recorded easily in the proper sequence on individual tracks. The resultant multitrack tape then can be played back, and the elements processed and mixed down to the final product.

Technical quality also can be maintained. The final mix from first-generation tracks, rather than a composite of multiple sound-on-sound dubs, will save 3 dB of additional tape noise from each of those transfer stages that would have appeared in the final product. Too many dubs or tape stages will make the final production sound "muddy," as each new layer of sound is added.

Advertisers might be tempted to buy your air time first if they know that the station can produce a high-quality professional commercial. Some stations put a restriction on commercials produced in-house, and do not allow airplay on other stations unless a production fee also is assessed before being aired elsewhere. This is done mostly for competitive reasons, although it could provide a nice source of second income.

"Spec Spots," or demonstration commercials, can be produced very quickly in an efficient multitrack production studio. Sales staff armed with these productions are sure to close more sales when the client hears how the product could be presented over the air. (If the client wants to hear additional, more complex production, get his advertising order first, or charge him a production fee so your valuable time won't be wasted; in most cases, the first spec spot produced will close the deal.)

A multitrack production studio is a tool that provides creative people with the ability to create more enjoyable and sellable product. *APfB* canvassed opinions from production staff working at a variety of West Coast radio stations, to see how multitrack recording and production techniques are being used.

The Stations

KDAY is a 50 kW AM station serving the black audience of Los Angeles, and has used multitrack production since chief engineer **Andy Laird** arrived 10 years ago. At that time a pair of two-track tape machines were installed in a studio wired for mono production. Material was recorded first on one track, then additional material would be laid down on the second track in sync with the first. Synchronization of the two tracks was maintained by monitoring the playback of the first track from the record rather than conventional replay head. The two completed tracks then were summed for a final mix on to tape cartridge for air play. KDAY's original two-track machines were Otari MX-7000s. Between 1975 and 1976 two MCI JH-110 two-tracks replaced the Otaris, and in 1977 a four-track, half-inch MCI JH-110 was purchased, opening the door for more exotic production. With AM Stereo coming soon to KDAY, the next addition may be an eight-track machine. The production studio currently is being converted to stereo while KDAY awaits the arrival of its Harris AM Stereo Exciter.

In-House Production

All stations *APfB* spoke with report greater creativity and operating ease with multitrack facilities. As KGB's writer/producer Tom Sarmiento says, "I produce a lot of spots where I do [many] character voices, and literally talk to myself back and forth. You can listen to yourself [on previously recorded tracks], and the timing is always right on. I just did a commercial for a computer company where I had pitch control on some of the subliminal voices that I used behind my own straight voice. Using an Eventide Harmonizer and some EQ gave a really neat effect. The client really liked it, and will go all over the market with it.

"We have concert promoters wanting their spots produced with all the updates for 'This coming Week,' 'This coming Friday,' 'Tomorrow Night,' etc. Multitrack makes it real easy for us to lay down the music track along with the updated voice tracks. It saves time."

But time savings is not the only factor in justifying a station outfitting a multitrack facility. According to KGB chief engineer John Barcroft, "It isn't so much that we're trying to save time in

KIIS-FM four-track production studio

every case . . . they can spend the same amount of time, and get twice as good a commercial. If someone says you've got to put all these effects together and synchronize it in two-track, the guy will just say 'the heck with it,' and not bother. In four-track, where it can be done easily, the guy tends to make better [sounding] commercials."

KMPC's eight-track facility also allows the staff to work with creative material from sources outside the station. A station's new jingle package has just been cut, and the mastering facility will provide KMPC with an 8-track submix from the 24-track master tapes, thereby allowing the station to do custom mixes for specific applications. The music and/or vocals can be separated, remixed in different proportions, or added to outside material for a variety of combinations.

KIIS receives a weekly syndicated program from Watermark Studios, North Hollywood, of which the voice track is that of an announcer working for a local competing station. As chief engineer Mike Callaghan puts it, "The last thing we need is to dilute our listener identity by having that voice on *our* station. Watermark sends us a two-track stereo tape without the voice. We dub it to the multitrack, and use a track for our own announcer to take his place. When played on our station you hear one of our jocks doing this incredible program. It gives the listener the feeling that we put the show together ourselves — not the syndicator."

The Los Angeles radio stations to whom we spoke do almost exclusively in-house production for themselves, and their advertisers. They generally do not rent for outside production, due to the abundance of independent recording studios in the area designed for this type of work. A client purchasing air time who does not already have a pre-produced commercial will receive a produced commercial from the station, usually at no additional charge. Generally this will be for airing only on that one particular station. Distribution outside the station may carry an extra production and talent fee.

KGB designed its studios to bring in outside clients. "The studios were designed and justified to be this complex," Barcroft says, "because we were going to bring in outside clients. Agencies are bringing their commercials in here and they're getting done for other stations . . . but only if they buy our air time."

The studios average a 40-hour work week. KGB runs a very heavy schedule though, starting at around 9 AM and continuing until 9 to 10 PM. Barcroft tells us that it is even difficult to schedule maintenance: "I've been trying to get in the [main multitrack studio] for two days to fix a pot — I finally snuck in while the guy was on his coffee break!"

Signal Routing

An in-house production studio must be efficient to operate. All of the facilities *APfB* visited are arranged for operator convenience, and employ audio consoles capable of working with multitrack. These consoles vary from the 20-by-8 at KMPC, to KDAY's 10-by-2 board. Each track from the multitrack tape machine appears on an individual

KFI also is a 50 kW Los Angeles AM station currently operating with the Harris AM Stereo system. The station's in-house production studio is an early Pacific Recorders turnkey package, originally installed in 1975 as a mono facility. Two years later an MCI JH-110 four-track, half-inch machine was installed, and in 1978 the existing console was converted to stereo with the addition of a second program and a second audition bus. The studio primarily serves KFI, but does see occassional use by sister station KOST-FM.

. . . continued on page 11 —

Solid State Logic

In the Foreground of Television Audio

Audio for video is on a lot of minds these days. Advanced video formats and transmission methods make dramatic improvement possible. Producers' concerns over the initial impact and residual value of their programmes make it desirable. EFP, new competitive arenas and increased consumer awareness make it necessary. And now, the SSL Stereo Video System makes it practical.

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And of course, the Solid State Logic Stereo Video System provides you with the ergonomic and sonic attributes which have made our companion SL 4000 E Series the leading choice of the world's great music studios.

Format Flexibility

The Stereo Video System's six bus mix matrix accommodates all audio-for-video formats. Along with standard mono, stereo and multi-track operations, each input may be panned between one of three stereo mix buses. This allows the engineer to freely divide the console into dialogue, music and effects sections as each project requires.

The Dialogue, Music and Effects mixes may be recorded in mono on a 3 stripe or 4 track, or in stereo on an 8 track or the multi-track master. Composite stereo and mono mixes of all 6 buses are derived from the master mix matrix for monitoring, transmission and/or simultaneous (first generation!) layback to the stereo video recorder. Alternatively, the six buses may be used for stereo mix and mix minus feeds during live coverage.

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Each I/O module contains an expander/gate, compressor/limiter, high and low pass filters, four band parametric equalisation, six cue/aux sends and tape electronics remotes. Master logic, push-button signal processor routing, patchfree audio subgrouping, and 8 VCA Group Masters ease complex productions, and always provide the minimum signal path.

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Complete details of all I/O module control settings are stored on floppy disc by SSL's Total Recall System, enabling console setups to be restored within .25dB accuracy. Not only does Total Recall save time on each production, it allows greater scheduling flexibility with fewer headaches than ever before possible.

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In live production, the SSL Real Time System enables complex sequences of all channel and group fades and cuts to be pre-programmed, and then manually executed with a single set of controls.

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The Solid State Logic Stereo Video System is available in studio and Outside Broadcast versions from 16 to 56 I/O modules, with up to 112 line and microphone inputs plus four stereo effects returns. Please call or write on your letterhead for complete details and prices.

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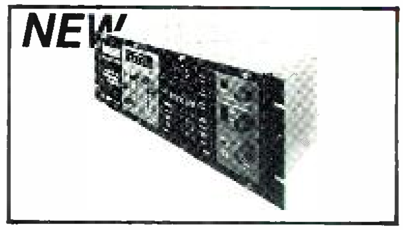

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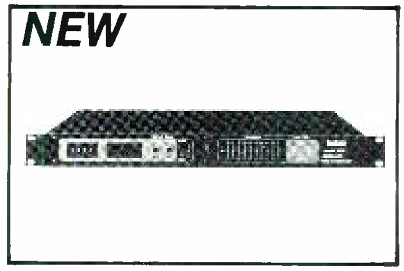

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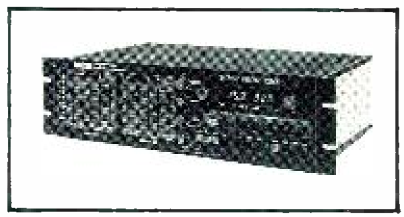

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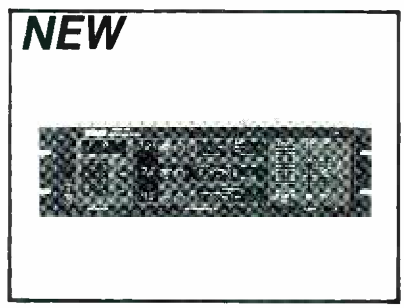

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KGB-FM four-track production studio, one of a pair of multitrack rooms available for in-house commercials and PSAs.

KGB-FM also has available a fully-equipped stereo production facility, which can work with a centrally-located announce booth.

— continued from page 6 . . .

input fader. At KMPC, these faders, along with those dedicated to other sources, can be assigned to any or all of the four submixer busses, which in turn will then be assigned to the desired program bus.

KDAY mixes its four-track into the audition bus, which then feeds the effects sidechain. All of the effects equipment is connected in series, with the final device output returning to the program bus through one of the input mixer channels. All of the individual effects device's inputs and outputs appear on the patchbay, should a reassignment, change of processing order, or total elimination of one or more devices be required.

KGB-FM, San Diego, is a top-rated AOR station. Its sister station, KCNN-AM, is all-news format, and carries much of the ABC Talkradio network, Cable News Network, and local news. The station completed its multitrack facility in March 1980. A full building wing away from the Air operations is devoted to two, nearly identical, four-track studios; an elaborate stereo studio; and any announce booth shared by one of the multitrack rooms, and the stereo studio. The only difference between the two multitrack rooms is the choice of the four-track machines, and a greater concentration of effects devices in one of the rooms. Three full-time writer/producers are employed by the station, along with a "mechanic" to assemble the final production and dub tapes.

KIQQ takes a slightly different approach to multitrack mixdown. Although all four tape tracks can be assigned to individual faders via a simple rotary switch selection, the station normally runs the four tracks into two pairs of stereo faders. Track #1 and #3 appear on a left and right input of one stereo fader, while track #2 and #4 appear on the left and right input of an adjacent stereo fader. Each fader has a balance control along with a rotary switch to select either left-only, right-only, or L+R to feed the left/right program bus. The multitrack outputs also appear on the remote input selectors feeding the last two console faders in a similar fashion.

By selecting the proper combination of left-only and right-only on the four faders, discrete mixing can be obtained. Also, this arrangement of two tracks into a stereo fader allows the level of stereo material, such as music beds recorded on the multitrack, to be controlled by a single fader. All inputs and outputs to and from the console appear on the patchbays, should a complete rearrangement of the console assign-

— continued overleaf . . .

KMPC, Los Angeles, is a 50 kW AM station playing music from the Forties, Fifties, and Sixties, and also has plans for AM Stereo. The station has been using an MCI JH-110 eight-track, one-inch machine since 1980. An impressive array of hardware complements the production studio: a Ward-Beck 20-by-8 console, UREI 839 Time Align monitors, and an AKG BX-10 reverb are just a few of the pieces of equipment giving this room the potential for creative quality production. Currently located in a temporary room, the studio will be moved to one with a more desirable length-to-width ratio; the moving project will be much easier than one might expect, since the installation is fully modular.

KIQQ's four-track production facility is equipped with Otari MTR-10 multitrack and stereo tape machines.

At the heart of the KIQQ studio is a Greg Labs 14-channel console, the last two inputs can be selected between six switch positions.

All Photography by Hank Landsberg

ments be required.

Track bouncing, or the combination of one or more tracks to a single or stereo pair of tracks on the same tape, was not often carried out. Four tracks seemed to handle most of the work load, and track bouncing was rarely required, or even feasible with so few tracks. A few productions though, occasionally were handled in this manner. The procedure is simple: lay three tracks, mix these to track #4, and now you can re-use tracks 1 thru 3. The final mix of all four tracks will be roughly equivalent to a seven-track production. It must be remembered, however, that during the re-recording of tracks 1, 2 and 3, track #4 must be monitored from the record/sync head for proper syncronization with the previously recorded track(s). Most 4-, 8-, and 16-track machines come equipped to accommodate such switching.

The bounced tracks will suffer a 3 dB degradation in signal-to-noise ratio but this was not considered severe enough, nor the process utilized often enough, to need additional noise reduction.

Mastering speeds varied from 7½ to 30 IPS. Working at 30 IPS KFI experiences a slight problem with machine start up time when atempting very tight cueing. Sometimes a verbal countdown would be recorded to backtime into the cue point, and the machine pre-rolled. KDAY has used 30 IPS to produce a nice delay effect by utilizing the time difference between the record and play heads. "This has the effect of 'fattening' voices on some production," says chief engineer Andy Laird. "The delay for a given tape machine can be calculated as follows:

$$D = (S/T) \times 1000$$

where, D = Delay (milliseconds)
S = Distance between heads (inches)
T = Tape speed (IPS)

The magnetic tapes used were always of the higher quality brands and type number. Flux levels generally were not elevated since there was a desire to retain the additional headroom, and it was felt that the small amount of noise sacrificed was of a minor concern when comparing the noise limits of the system and listener environment against the much more audible effects of tape saturation and clipping. Non-elevated levels also provided operators with a little safety pad against distortion should levels hit the meter pins. However, KIQQ does use elevated levels relative to the 185 nWb/m standard of +6 dB on the four-track machine, and +3 dB on the two-track machines. Scotch 250 and 206 tape is used respectively on these machines. Careful level control is maintained by allowing only the production director, Bob Sky, access to KIQQ's multitrack production studio.

In most of the studios visited, patch bays gave access to all inputs and outputs. Two exceptions were found at KMPC and KFI. Here, routing switchers were employed to direct signals and effects to their proper assignments. Both two stations did have patchbays, but it was not necessary to use them in the daily operation. The patchbay at KFI was buried down low in the cabinetry, and accessible only to engineering personnel for equipment testing, while KMPC's patchbay appeared in the same rack with the routing switcher. As chief engineer Steve Colley told us, "In any facility that you have to patch daily . . . there's something wrong with that facility. We have a few cords [in the bay] at the moment for custom set up to patch around things that are out for service. The intent though is not to patch for things we normally need."

The patch bays at KIQQ and KGB are wired with equipment outputs on the top row, and equipment inputs along the bottom row. Normals are from top to bottom. When a patch is inserted in the top row it does not break the normal. In this way an output can be distributed to another input without disturbing its normal feed. A row of bays wired in parallel provides a "straight wire" distribution system to obtain multiple feeds from one source. Since most of the equipment has been designed with very low output impedence and high impedence inputs, fanning the outputs does not present a problem.

Monitor Systems

Monitoring speakers among the sta-

A MCI JH-110 four-track and Pacific Recorders stereo console form the basis of KFI's in-house production studio

KIIS-FM and sister station **KPRZ-AM** are located 19 floors above Hollywood on busy Sunset Boulevard. The AM station competes with KMPC, while the FM programs hits from the Top 40. The in-house multitrack room was built in 1975 by Westlake Audio, Los Angeles, using an auditronics "Grandson" console. A Scully 280 four-track, half-inch tape machine was obtained from nearby Capitol Records, and restored to mint condition. According to chief engineer **Mike Callaghan,** the KISS installation was considered unique because "at that time Westlake Audio had a policy that when you bought a console of that price, they would do the wiring for you." Westlake's multitrack experience was a great benefit to KIIS, in setting up the station's in-house broadcast production studio.

tions varied in size from JBL 4311 to UREI 839 Time Align units. Small Auratone 4C Sound Cube monitors also were found at KGB for, as producer Tom Sarmiento says, "mixing as if you were in the car." KDAY's RSL 3600 monitors were equalized for flat response at the mixing position using the UREI Soni-Pulse system. KMPC places its UREI 839 monitors very close to the mixing position. They are located only a few feet directly in front of the operator, and hung from the ceiling. Placement was chosen more by the carpenters, who were concerned with structural support, rather than for a "near-field" monitoring effect.

How well do these monitors represent the final desired sound that would come out over the air on a small radio speaker, we asked? Had any consideraiton been given to employing small speakers for the final mix? KMPC's chief engineer Steve Colley says that, since the quality of the average AM radio and listening environment is consistently bad and unpredictable, it is better to strive for the highest quality sound and good performance on a wide range of receivers. "Who is to say that what we come up with for a three-inch speaker would sound at all like the listener's." he offers. "I've made the mistake of making the radio station's processing sound good on my car radio, and *not* necessarily on someone else's."

Whichever monitoring system is choosen, it should be capable of providing a final mix that will sound good to an audience in *their* listening environment over large or small monitors, and various receiver qualities.

Outboard Effects

Effects devices at most of the stations consisted of limiters, compressors, equalizers (parametric and graphic), pitch shifters, reverbs, etc. At KGB we discovered a small portable flanger that originally was designed for musical stage use. Although it looked a little out of place among the professional gear, the unit did provide an acceptable flanging effect, and will continue in service until replaced by a studio-oriented item.

A budget-minded station wishing to add effects to a production facility might do well to shop in the local musician's outlet. A wide variety of effects devices are available at low cost, and provide an excellent method of adding an extra "touch" to production.

Each station was asked in what way it would like to improve the quality of the in-house facilities. Surprisingly, most were pretty happy with their current studios. KGB wants a digital delay and a Vocoder to add an even greater variety of special effects. KDAY will add a multitrack headphone monitor mixdown panel, so that staff can monitor or audition multitrack material without having to assign it to a program bus. KIIS might relocate a few pieces of equipment to bring controls just a bit closer to the operator. KIQQ hopes to budget for a larger console, since its present model is filled to the maximum with assignments, and for more multitrack tape machines to add greater versatility to the room. KFI would like a console with a few more recording studio features, such as EQ in each channel module. And KMPC would like an even larger console with more inputs and faders, stereo synthesizer, four more cartridge tape machines, a second set of monitors to simultaneously monitor the audition channel, possibly a third two-track machine, and a second turntable for "on-air" style operation.

If a station is considering its first venture into the world of multitrack production, it is possible to "test the water" with a small expenditure by simply adding a semi-professional four-track, quarter-inch tape machine to its existing mono or stereo production room. The machine should be chosen that has the ability to monitor from the record head for synchronous operation. Four console faders should be dedicated for mixdown, or an outboard passive resistive mixer constructed.

A station wishing to go beyond four-track may want to consider some of the semi-pro machines with eight tracks on half-inch tape, such as the TEAC Tacam 80-8, and Otari MX-5050. The available quality of such equipment will be more than adequate for most applications. Should a station feel the need for an even greater number of tracks, it could be helpful to visit a local recording studio to learn from its experience with various equipment and track formats.

* * *

Multitrack production techniques can add a sparkle to a station's commercials, promotions, and pre-produced programming that is certain to draw greater attention from its audience and advertisers. The basic concepts of multitrack are not difficult to master. If you ask "How am I going to get more creative work out of my disc jockeys with multitrack?" I can only say that without giving them this opportunity their talents will never be discovered at your station! ■■■

KIQQ, Los Angeles, is the most recent station to employ multitrack in a newly completed four-track studio. An Otari MTR-10 half-inch tape machine was chosen by chief engineer **Lyle Henry**, because "we like to try new product, and the factory will bend over backward to support us." A pair of two-track Otari MTR-10 machines also serve this room. Consistency of product is assured by the fact that only production director Bob Sky is allowed access; the room is locked outside of his working hours. Other station talent provides voice work, and have 24-hour access to two other two-track studios.

HIGH-QUALITY AUDIO FOR VIDEO PRODUCTION

A Case Example at BROADWAY VIDEO, New York City

by Paul D. Lehrman

Now that cable, satellites, and FM-stereo simulcasts are having an increasing impact on television production, the sophistication of the medium's audience is growing. At the same time, steadfast music listeners of the past are looking towards new forms of entertainment, including video. It's only logical, then, that the fields of audio and video production and engineering are beginning to merge. To the "traditional" recording studio wanting to become more involved with video, such a progression can represent serious pitfalls — the putting together of a sweetening or post-production facility involves a lot more than just buying a ¾-inch U-Matic videocassette machine, and a timecode reader. Conversely, established TV production houses, with their long-time orientation towards picture and picture alone, are often slow to realize and act on the pressing need for high-quality audio for video.

The United States, for many reasons, lags behind Japan and much of Europe in improving the quality of television sound. The FCC has been dragging its heels about Stereo TV which, if and when it is allowed to happen, can't help but increase audience awareness of television sound. On the other hand, Japanese broadcasters, who already are heavily committed to Stereo TV at home, are beginning to market component video systems to consumers in the US that contain at least medium-fidelity audio componentry. The television networks, too, have stopped sending the audio portion of their programming over Telco landlines, and instead are routing it via satellites along with the video signal.

But, at the forefront of improving sound quality for TV and video production are the independent post-production facilities that incorporate into their operation state-of-the-art audio technology; a good example of which is New York's Broadway Video.

Broadway Video, the brainchild of Lorne Michaels, creator and former producer of *Saturday Night Live*, lays claim to quite a few industry "firsts" and "onlys." Opened with one room in the Fall of 1979, during Michaels' last season with the show, one of their first assignments was the re-assembling of *SNL* segments into prime-time NBC specials. Broadway was the first post-production house in New York to be built completely around one-inch C-Format videotape machines. The facility now includes three on-line editing rooms, the largest of which is reported to be the first video facility in New York to use UREI Time-Aligned monitor loudspeakers, and lays claim to being the only video room on the East coast to feature a certified Live-End/Dead-End control area.

With two entire floors and sections of two others in the historic Brill Building, Broadway Video today also contains an off-line ¾-inch editing room; a music recording studio complete with a fully-equipped New England Digital Synclavier digital synthesizer; a computerized graphics room; film cutting rooms; a four-channel screening room the facility shares with Paul Simon's production company; plus plenty of office space. Also in the works is a Foley stage.

In addition to no fewer than 14 Sony one-inch video recorders, four Sony computer-controlled video editors, and enough BTX Shadow SMPTE synchronization equipment to please just about everyone, Broadway is the proud owner of a 16-track Otari MTR-90 (soon to be replaced with a newer model 24-track, designed, with Broadway video's help, expressly for post-production), Otari MTR-10 and MX-5050 four- and two-tracks, as well as Studer and MCI transports, several Sound Workshop Series 30 audio consoles and, in the large editing room, a 28-input Series 40 console and an MCI automation system, custom designed to interface with the video editing computer.

All photography by Tall Walton

Audio Emphasis

Michael Werner, Broadway Video's vice-president for technical operations, and who has been with the company since the beginning, describes himself as "an old audio man — not to mention a hi-fi nut," and considers that he has been the chief influence in keeping the company's audio capabilities on a par with the video side.

"The emphasis here is equally on audio and video," he offers, a principle that covers both equipment and staff. Within the 12-person technical and editing crew are four audio-only engineers, including a new maintenance engineer who was hired away from New York's Electric Lady Studios. Other members of the crew have extensive experience in major recording studios, or with sound-reinforcement companies. "You have to have hip sound people on the staff," says Werner. "For one thing, we rent a lot of equipment, and if the rental companies think you're a video house,

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they'll send you garbage. We have people here who can work on [the rented equipment], and send it back in better shape than it came in."

Even some of the video staff at Broadway come from audio backgrounds — as far as Werner is concerned, knowing audio and learning video makes just as much sense as doing it the other way around. "The compentence of our personnel with sound is high on our list of qualifications," he says.

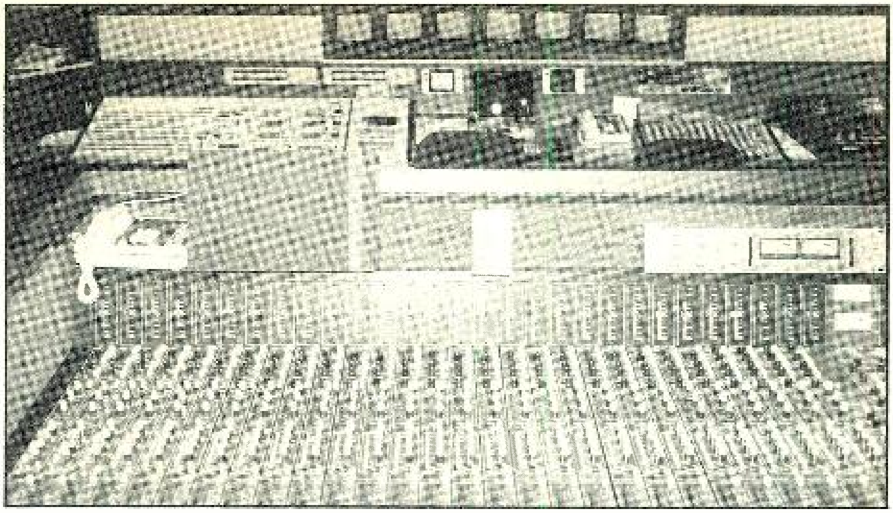
Monitoring: Having the right equipment, Werner considers, starts with the monitoring system. In ER-3, Broadway Video's large editing room, the front half of the space houses the video equipment, as well as a small Sound Workshop audio console, and a pair of UREI 811 monitor speakers. The rear half, raised on a platform, is the LEDE area, with a Sound Workshop Series 40 automated board, and another pair of UREI speakers. A transparent plastic "shield" drops down a foot or so from the ceiling to provide acoustic separation between the two spaces, while the free air underneath the shield functions as the "Dead End" of the audio area.

Because the product coming from the room may end up going out over a wide variety of media — including broadcast television, cable TV with (mono) satellite links, FM stereo broadcasts or simulcasts with landline links, professional and consumer video tapes and disks, and even record albums (Werner considers his room quite capable of that task) — there is a wide choice of monitor speakers, including JBLs, Auratones, Dahlquists and, of course, the speakers built into Sony and RCA consumer TV sets.

"We checked the room against the [Eastlake] room at Soundmixers studios [located on the second floor of the same building], and it transposes well," says Werner. "It also works just fine with Altec 604Es and Big Reds."

Playing with the signal path is important too, and Broadway Video has several monitor amplifiers available, ranging from tube McIntosh models to "junk," low-fi units. One option that the company often uses is to modulate the audio on to an actual television signal, and record it on a Betamax cassette to see how it comes across in that format. In some ways, Werner asserts, "You have to use the same kind of tricks you did in 1965, when the 45 single was the most important product."

Recording, processing, and control: Otari tape decks are used because, as Werner puts it, "they think they're VTRs; they back up and stop nicely, and always park accurately and quickly." There's also a wide selection of audio processing gear from Eventide, Orban, Lexicon, Ursa Major, Audio + Design Recording, AKG, and other pro-audio manufacturers. Most recording is done with Dolby A noise reduction. Everything is used: "We're lucky we don't have a single piece of equipment that lies fallow," says Werner.



Sound Workshop Series 40 in Broadway's Edit Room #3

"We have things that can do such marvelous stuff," he notes, "that no one person knows them all well enough to be an expert. I don't know anyone [on our staff], for example, who fully knows how to drive a Lexicon 224X digital reverberation system." Where he sees current audio technology falling down, however, is in the area of control.

"All of the major video editors have general-purpose interfaces," he explains, "which can activate contact closures according to timecode cues. Some of our equipment, like the editors and switchers, can even talk to each other using high-level computer protocols. But audio control technology is about four years behind video. Audio manufacturers are beginning to realize that they have to get on with the development of new control schemes, so that we have some way of automating them, which is really the only way we will be able to take full advantage of their capabilities. Of course, it's going to be expensive. Putting a timecode cue system in a Lexicon 224X, for example, which is already a pretty heavy investment for the average studio, could drive the price beyond $20,000.

"Miniaturization has helped — it allows you to pack more smarts into equipment. And we can do some stuff here that the manufacturers can't. We're working on a general purpose controller for all of the gizmos we have lying around, but it's going to take a while, and we have no plans to market it. There's no standard out there — although with a computer like a Hewlett-Packard HP-85 there could be — so it wouldn't make any sense to try to sell it to somebody else."

Control Alternatives: Until full automation of processing gear can be accomplished, Broadway Video relies on two ways of establishing timecode-based access to effects. One is to use the automation system fitted to the facility's mixing consoles. With the plethora of reverb units and other outboard devices at hand, effects can be preset through separate devices which can then be called up, crossfaded, or muted through the console. Although there are sufficient two-track machines around the facility to do multiple-machine mixes, Werner says that the preferred method is to record and manipulate the effects on a single piece of multitrack tape. This technique reduces generation noise, but does have drawbacks: it doesn't allow for random access, and if a cue or an effect needs to be moved slightly in time, it is difficult to accomplish.

To overcome that drawback, Broadway Video relies heavily on its other method for cueing sound from code: its in-house Synclavier digital synthesizer. Not only is the Synclavier considered by many to be a very versatile synthesizer, it also has built-in 16-track digital recording facilities, both for internally generated sounds and for organic sounds and even dialog, which can be entered through its sampling circuitry. All of the recorded information (up to 75 minutes worth) can be manipulated in all domains, including time, by the unit's computer processor. "It can store an event in RAM," explains Werner, "simultaneously look ahead to the next event, and then find it."

By itself, the Synclavier cannot read SMPTE timecode, but it can be synched to pulses coming from an audio tape that also contains a code track. Broadway Video routinely uses the Synclavier for sound effects, and Howard Shore, former music director of *Saturday Night Live*, recently composed the score for horror-film director David Cronenberg's new feature film *Videodrome*, entirely on the synthesizer.

Audio Production Flexibility

Working on projects like the Simon & Garfunkel Concert in Central Park involves every facility that Broadway Video has to offer. The concert was taped for broadcast over HBO (with mono audio), for FM-stereo simulcast over a 50-city, landline-linked network, and also for a two-record Warner Bros. record album. In addition, the concert is being prepared for release on both videocassette and videodisk release. All

MODULAR PROCESSING

EQF-2 *$449.00*

The EQF-2 combines a 3-band sweep equalizer with a sweep Hi and Lo pass filter section. The EQ has switchable peak/shelf on the Hi and Lo sections, and reciprocal 12 dB of cut and boost on all sections. The filters are second order Butterworth and can be switched separately from the EQ section.

SPECIFICATIONS

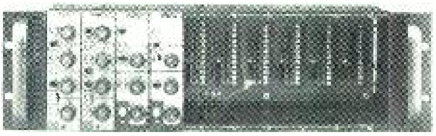
FREQ. RESPONSE:	*± 1dB 20 Hz – 20 kHz all sections in*
THD & IMD:	*Below 0.1% at max. I/O*
NOISE:	*–123 dB below max. I/O*
FILTERS:	*Hi pass 20-500 Hz* *Lo pass 1-20 kHz*
EQ LOW:	*25 – 500 Hz*
MID:	*250 – 5kHz*
HI:	*1 – 20kHz*
MAX. I/O:	*+20dBm with optional Jensen xfrmr*
SIZE:	*1-1/2" x 5-1/4" x 6"* *(industry standard)*
WEIGHT:	*2 lbs.*

CX-1 *$449.00*

The CX-1 is a very versatile module combining a "soft knee" compressor/limiter with a switchable expander/gate. The CX-1 uses the proprietary Aphex VCA chip to provide an extremely clean overall sound. The expander is adjustable from 0 to 100 dB of expansion (gating) and is the *only* noise gate on the market that can be guaranteed not to click or pop. The unit features a multi-functional LED display that indicates input, output, compression or expansion levels.

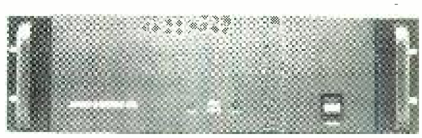
SPECIFICATIONS

BANDWIDTH:	*± 1dB 20 – 20 KHz all sections*
THD, IMD:	*Less than 0.2% at max I/O*
NOISE:	*–85 dBm*
MAX I/O:	*+20 dBm (+30 dBm with optional Jensen xfrmr)*
SIZE:	*1-1/2" x 5-1/4" x 6"* *(industry standard)*
WEIGHT:	*2 lbs.*



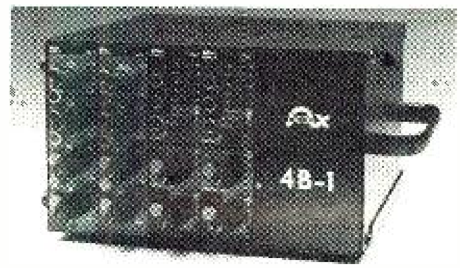
R-1 *$195.00*

The R-1 holds 10 Aphex modules and provides barrier strip access to all inputs and outputs. Power and ground are bussed.



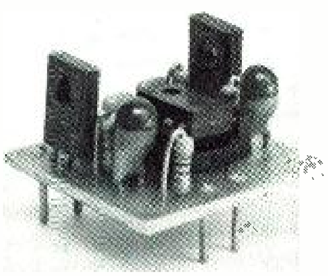
PS-1 *$275.00*

The PS-1 is a ± 16V @ 3.4A regulated supply with OVP that will power two R–1 racks.



4B-1 *$349.00*

Self-powered, the 4B-1 is for the mobile engineer. It holds 4 Aphex modules and has a built-in patch board on the rear with 1/4" and T-T size jacks.



2521 – OPERATIONAL MODULE
$35.00 (singles)

The 2521 Operational Module is a high speed, high output, short circuit proof buffer that takes on the characteristics of the IC that is plugged into it. It is current limited and can put out a full watt of power into a 62 ohm load.

The 2521 output transistors have a 3 amp rating for superior reliability. The unit is also 100% field repairable, so there's never a need to discard a complete module because of a defective 10¢ resistor. The 2521 can be continually and easily updated to meet changing needs.

FEATURES

100% Field-repairable
100% short circuit proof
Greatly improved overload characteristics
Built-in power decoupling
Socketed IC eliminates obsolescence
Extremely low noise current

SPECIFICATIONS

BANDWIDTH:	*4MHz*
THD (at clipping –1 dB):	*0.02%*
IMD:	*0.02%*
GAIN:	*50,000 Min.*
SLEW RATE:	*>10 v/μ Sec.*
OUTPUT NOISE:	*–113 dBm*
MAXIMUM INPUT:	*30 Volts P-P*
MAXIMUM POWER OUTPUT:	*1 Watt (+30 dBm)*
MAXIMUM VOLTS OUTPUT:	*Supply –4 volts P-P*
MAX. SUPPLY VOLTAGE:	*±18 volts (with LF 351)*

**High voltage, high output versions are available. Consult the factory for details.*

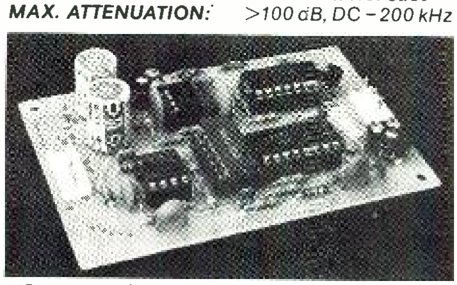
VCA PRODUCTS

1537A VCA IC *$10.50 (100's)*

The 1537A is the *only* monolithic Class A voltage-controlled attenuator on the market today. Its patented design features extremely low distortion, low noise, high stability and wide dynamic range. It can provide more than 100dB of attenuation at +20 dBm. Its high slew rate gives low T.I.M. and makes it useable from DC to 50 MHz.

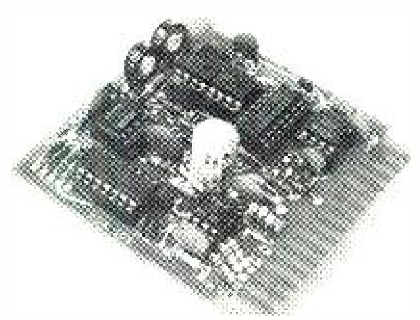
SPECIFICATIONS

BANDWIDTH:	*DC to 50 MHz*
THD:	*0.004% TYP*
IMD:	*0.03% TYP*
NOISE:	*– 90 dBv worst case*
MAX. ATTENUATION:	*>100 dB, DC – 200 kHz*



VCA 500A *$89.00 (singles)*

The new VCA 500 A utilizes a 1537A VCA IC to significantly improve the performance and overall sound quality of the MCI JH-500 series console. Conversion takes only a few minutes per channel with plug-in convenience.



VCA 505 *$89.00 (singles)*

The VCA 505 is an expanded version of the highly-acclaimed 1537A Voltage Controlled Attenuator. It utilizes a 15-pin card edge mount package for easy installation, has multiple bufferred control inputs for maximum versatility, and requires no additional circuitry.

SIZE: ***2.75" high x 2.85" deep x .72" wide***

APHEX SYSTEMS LTD. 7801 Melrose Avenue · Los Angeles, California 90046 · (213) 655-1411 · TWX: 910-321-5762

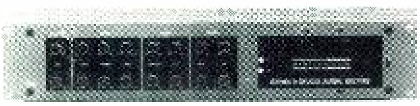
INNOVATIVE PRODUCTS FROM APHEX, FOR SUPERIOR SOUND REPRODUCTION

The remarkable Aphex Aural Exciter is a unique proprietary audio processing device that makes use of highly advanced psychoacoustic principals to effectively restore and enhance audio presence, brightness and intelligibility. The patented psychoacoustic process creates the perception of an increase in mid and high frequency energy, with *no* actual increase in power or level.

The Aural Exciter can produce dramatically improved clarity, dimension and character in any sound system or application. It can also reduce distortion in P.A. and sound reinforcement applications by providing increased penetration and audibility at reduced power levels. The device can be added to virtually any new or existing system with no danger of overloading other components or triggering compressors or limiters.

The Aural Exciter is a single-ended process, requiring no decoder. Once encoded, copies made from a processed tape sound every bit as good as the original.

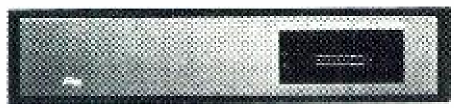
The Aphex Aural Exciter is available in three models, each is specially designed for a specific application.



APHEX II-S $2,950.00

The Studio Aural Exciter is engineered for the sophisticated recording and production studio, as well as advanced sound reinforcement applications. In the studio, the Aural Exciter effectively restores the presence and clarity which the recording process removes, reviving that bright, unmistakable "live" quality. It can also make certain segments "stand out" without actually being louder. Used typically in stereo mixdown situations, this latest version of the Aural Exciter features increased flexibility so it's ideal for virtually all types of program material, from the hardest rock and roll, to the subtlest movie dialogue and sound effects.

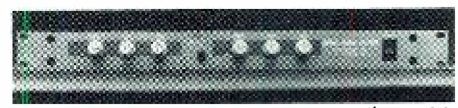
The Aural Exciter is also well suited to stage and concert use. It can make any P.A. system sound much cleaner, brighter and intelligible without adding any level or feedback to the house or monitor system. It is particularly effective in filling acoustic spaces to eliminate dead spots. The device cleans up sound in overly reverberant halls and makes speaker location much less critical.



APHEX II-B $2,950.00

The Broadcast Aural Exciter has all the remarkable features and capabilities of the Studio unit, plus complete R.F. shielding and safety bypass relays in the event of power failure. Designed specifically for on-air use, this unit provides AM stations with the clarity and brightness of FM, while restoring to FM the naturalness and openess normally lost due to processing.

The most impressive aspect of the Aphex Broadcast Aural Exciter is the fact that the lower the quality of the playback system, the better the comparative benefit derived. The sound of your broadcast will satisfy the most demanding audiophile, and at the same time grab the attention of the rush-hour commuter.



APHEX AURAL EXCITER TYPE B $495.00

The Aural Exciter Type B is engineered for less demanding situations. It utilizes the same psychoacoustic principles to make Aural Excitement available to small clubs, studios, halls, restaurants, musicians, tape duplicators and sound contractors operating on a more modest budget. Retaining the most important features of its bigger brothers, the Aural Exciter Type B is a small, lightweight package with extensive capabilities limited only by the user's imagination.

SPECIFICATIONS APHEX II

FREQ. RESPONSE:	*15HZ-50KHZ +0-.2 dB*
THD, IMD:	*.05% at max I/O*
NOISE:	*110dB below max output*
CROSSTALK:	*Better than -80dB*
MAX I/O (with standard Jensen output xfrmr):	*Selectable +21dB, +24Bm, + a user definable position*
INPUT IMPEDANCE:	*Selectable 600ohm or bridging, 40K Bal, 60K unbal*
OUTPUT IMPEDANCE:	*50ohm*
METER:	*2 color, 2 channel VTF with LED legends for drive, limit, peak and meter status indication*
SIZE:	*3-1/2" x 19" x 9"*
WEIGHT:	*19 lbs.*
POWER REQUIRED:	*100-240 VAC 50-60 Hz*

**Single-ended transformerless and balanced transformerless outputs optional.*

SPECIFICATIONS - TYPE B

FREQ. RESPONSE:	*10 HZ-100K HZ ± .05 dB*
THD:	*Less than .01%*
NOISE:	*-90 dBV*
OPERATING LEVEL:	*Selectable -10 or 0 dBm*
MAX I/O:	*+20 dBm*
INPUT IMPEDANCE:	*47K ohm unbalanced*
OUTPUT IMPEDANCE:	*150 ohm unbalanced*
METER:	*Tri-colored LED for drive level*
SIZE:	*1-3/4" x 19" x 6"*
WEIGHT:	*4.5 lbs.*
POWER REQUIREMENT:	*100-130 VAC 50-60Hz (export version available)*

COMPELLOR™ COMPRESSOR/LEVELER/PEAK LIMITER $995.00

The Compellor™ is a unique, revolutionary audio processing tool that combines the functions of a fast compressor with slow gain riding and an overall peak limiter. It provides complete flexibility in dynamics control when used as a broadcast pre-processor, as well as in the recording studio or live p.a. situation. The resulting sound is smooth and dense with an increase in perceived loudness and brightness.

The variable slope compressor operates over a 30 dB range with attack and release times controlled by program dynamics, eliminating "pumping" and the choked sound associated with deep compression.

Audio leveling over a 20 dB range maintains the audio in the "knee" of the compressor providing a uniquely dynamic compression which is rich in transient quality and openness, with an absolute ceiling maintained by the peak limiter.

The balance between compression and leveling actions is continuously variable; adapting the Compellor™ and its effects to an enormous variety of material.

APHEX SYSTEMS LTD. 7801 Melrose Avenue · Los Angeles, California 90046 · (213) 655-1411 · TWX: 910-321-5762

Anyone Could Put the Phone Jack in the Correct Place...

The UREI Broadcast Consoles

At UREI, we believe details and solid construction make the difference between a good product and an excellent one. For instance, headphone cords are on the left side. But virtually all on-air boards put the jack on the right side of the console. On the new UREI boards, we give you two jacks, one on the left, where it should be, and another on the right to let two people monitor simultaneously.

Before we started designing the new Series 1650 and 1680 consoles, we talked to you . . . engineers, jocks, announcers and consultants. Then we went to the drawing boards to plan a line of on-air boards with the features you wanted.

The result is a fresh answer to the operational requirements of the broadcaster of the '80's. Six all-new on-air boards . . . choice of 5 mixers (1650 Series) or 8 mixers (1680 Series) . . . choice of Penny & Giles slide pots, Shallco or conductive plastic rotary attenuators.

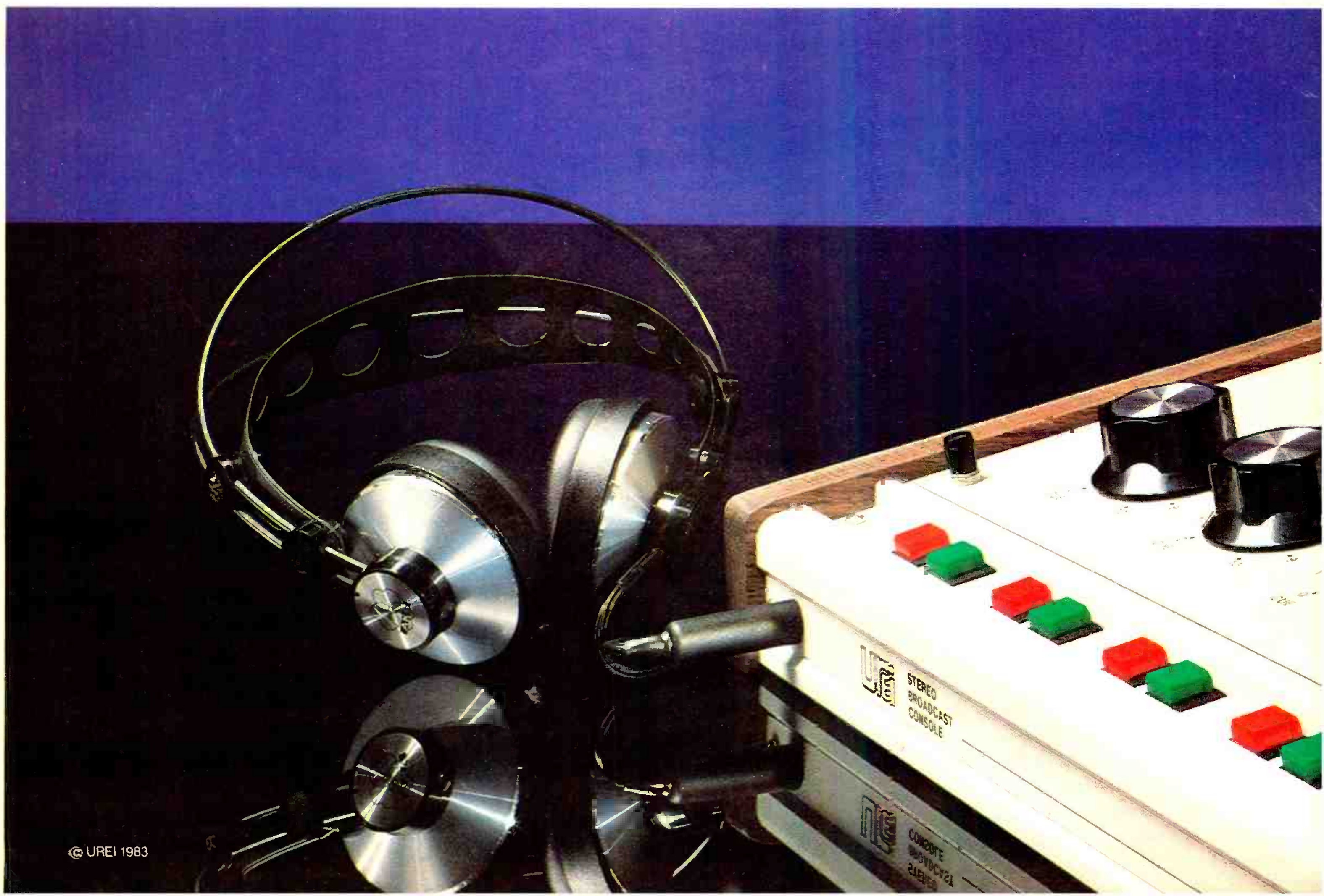
High Output, Low Distortion, Low Noise

The Series 1650 and 1680 Consoles can easily handle today's dynamic program material with +24 dBm output into 600 ohms. Signal to noise is better than 90 dB. At normal operating levels of +4 to +8 dBm, THD is typically less than 0.02%.

Flexible, Simple Operation

Each mixer position has two inputs selected by a gold contact rocker switch. Two banks of four push-buttons each may be assigned to any mixer input for additional sources such as remote or network feeds.

All channel on/off switching is performed by reliable FET switches; extra contacts are provided for activating cart machines or turntables.



avoid high-frequency loss, and skewing."

New cart machine technology has helped to eliminate the problem of carts requiring subtly different azimuth alignment, Brown considers. "We use the ITC Series 99B, which does everything except the windows! It has what's called an ELSA Function, which means that it erases, locates the splice, and adjusts head azimuth. In days gone by, you'd have to erase a cart by hand, and it wouldn't be a very good job because everybody had a different technique of bulk erasing. Then the machine lays down a tone, and aligns the record-head azimuth for the optimum output on this particular cartridge. It then erases that tone, and locates the splice. I then roll past the splice about three seconds at normal speed to stabilize the tape. Then you hit record, and you're ready to go.

"Every playback machine in the station is aligned with the same alignment cart, so they're *all* on the same standard."

Laird favors alignment cartridges from Fidelipac, or Standard Tape Laboratories. "I don't see any problem with having an alignment tape on a different type of cart than your programming, so long as the carts are consistent through the years. The cart from Fidelipac is an old-style cart with an adjustable post, and they tell me they do it that way because they can get exactly perfect consistency over years of manufacturing alignment tapes. But that alignment cart is going to play back in a certain way. If you adjust the playback head so that it's perfect, when you put in a different type of cart, and the record head is adjusted so that it plays back properly, it should be proper throughout the whole radio station."

Laird also offers a word of advice on the selection of the cart machine: "First thing is [to make sure] that a machine won't stop, drag or jam, so you've got to watch for reliability. Then you go for the best audio quality: lowest wow and flutter, and the highest frequency response. Audio circuitry has improved incredibly in the past few years, and some of the new high slew rate ICs are vastly better than anything we've ever had in broadcast audio. Two brands of cart machines have redesigned their circuitry with the new ICs in mind — I'm partial to the ITC Series 99, but I've also done some work with TomCats out of Pacific Recorders; they're a great sounding machine, no question about it."

Phase and azimuth are of partiular concern to Schulke and TM Programming, and their reel-to-reel release formats. Recording duties at the two companies are handled by Studer and MCI machines, respectively. The emphasis on alignment, azimuth, and phase may be even more important, however, since neither firm has control of a station's playback decks.

"Most of the radio stations in the country have marginal audio gear," says TM's Pat Hogan. "Usually it's the major market stations that have the best equipment. It's rare that the small to medium market stations have the state-of-the-art equipment that we or Schulke have. That's why we both have such tight, rigorous phase standards. We like to think that if we sent out a tape that has no more than 12 degrees phase error at 10 kHz, that will allow the radio station to have up to 45 degrees phase error. Since HF phase error is inaudible until 60 degrees, they'll still have a clean sound."

To assist their client stations in maintaining phase and azimuth alignment, TM Programming distributes alignment tapes produced for them by Standard Tape Laboratories on Ampex 642 tape for a standardized playback adjustment.

"Also," says Hogan, "a lot of syndicators will lay down a newly generated tone when they duplicate their tapes. We don't think that's a true representation, because it's *after* the fact. We will put azimuth tones on the master tape itself for duplication.

Meticulous, scheduled maintenance was emphasized more than once as being absolutely necessary if a station is to consistently put out a quality sound. "At Schulke," says Joel, "each transfer engineer assumes that the room doesn't work; they put it through the paces *every* shift. There is an extensive checkout sheet: de-magnetizing is done before each reel of tape; and azimuth and bias are set for that reel of tape. We also record something called 'dailies' — reference material that is laid down on tape every day so that we can go back a week, two weeks, or a month later and listen to the same piece of music and compare. If we have even a slight erosion [in sound quality], we can detect it.

"Additionally, we work very closely with all our [client] radio stations. I'll go out in the field with recommendations on how long playback heads should stay in serivce, and procedures for setting up the playback machines."

"We routine maintenance our cart machines here weekly," says Laird of KDAY. "You can actually have done your cleaning and demagnetizing and check your alignment, level, and frequency response, and be out of the studio in an hour. Anything else, wow and flutter, or something involved, I do in the shop. When you design your studio, set it up so you can take the gear out and work on it in the shop where you have what you need."

The major point from all those we spoke to was simplicity in all phases of the operation.

"The big problem," Laird concludes, "is getting the sound off the record flat and on to tape. If you keep the electronics as *simple* as possible, you have much fewer possibilities for things to go wrong, or for audio degradation due to the electronics." ■■■

the NAB EQ curve used in recording. A meter that indicates the amount of level going on to a tape after EQ is the most *valid* way of recording. You also have the playback meter so that you can get the replay level from event to event set to the same level.

"To this end, you should monitor the playback level of a stereo cart with a mono meter, adding both the left and right channel together. That's the way the audio processing gear sees the level, and mostly the way your ear hears it. If I have a left channel-only event, it will actually record on to the cart 6 dB hotter than a monaural sound of the same thing, if you're looking at your mono meter. So, if I have left channel-only, it will read -6 dB, instead of zero. The tendency would be to increase the recording level so that the cart plays back properly at the same level in mono."

Tape Selection

The choice of audio monitoring systems cuts across the spectrum of available product, as does the selection of recording tape.

"We're using 3M 250 for mastering," says Schulke's Irv Joel, "and Ampex 632 for duplicating. The 632 is a good, low-noise tape with consistently good slitting and magnetic properties, and it stands up well in the field. We're always investigating other people's formulas, but we're trying to fit a piece of material to a system, and this works best for us. Additionally, when we purchase tape, we have input quality control on the raw material, electrical and mechanical tests, before we decide to use it."

"We're always experimenting with different types of duplication tape," says Pat Hogan at TM Programming, "and through engineering director Chuck Webster's experience, and our efforts with tape companies, Ampex established the 642 tape as a product line. We have a thorough system of checking for bad tape, breaking it down into shipments, and lot and batch numbers. We can isolate the bad batch, and ship it back."

"What we're using," says KRLA's Douglas Brown, "are Capitol Audiopack cartridges, AA3s with HOLN-Q17 tape. We particularly like them because the noise floor not only measures well, but they also sound quieter than the others."

"For FM radio," advises Laird, "wind your carts with some kind of high-level recording tape, but *don't* use the extra headroom for more level. Use it for *transients*. One of the real audible areas of degredation is headroom, so I prefer to use the new tapes to gain headroom, as opposed to gaining signal-to-noise. [For FM] I'm a real fan of closed-system noise reduction. It can be a real maintenance headache; it takes a lot of care, and a specific cart machine to do it properly, but with careful maintenance a dbx system can sound really good on the air."

At KRLA, however, Brown finds maintenance problems enough to cause him to shy away from the use of noise reduction in carting music. "From that, and what it adds to the sound, I don't think it's worth the addition of all that hardware," he considers. "We're talking about the broadcast chain, and there is inherent noise in transmitters and receivers. I don't think [noise reduction] is particularly realistic in that context. The higher the level that you get on the tape, the less noise problem you have on the catridge."

"Rather than use noise reduction," says TM's Hogan, "we find that by cleaning up the audio chain, and removing as many transformers as possible, we're going to get a clean sound."

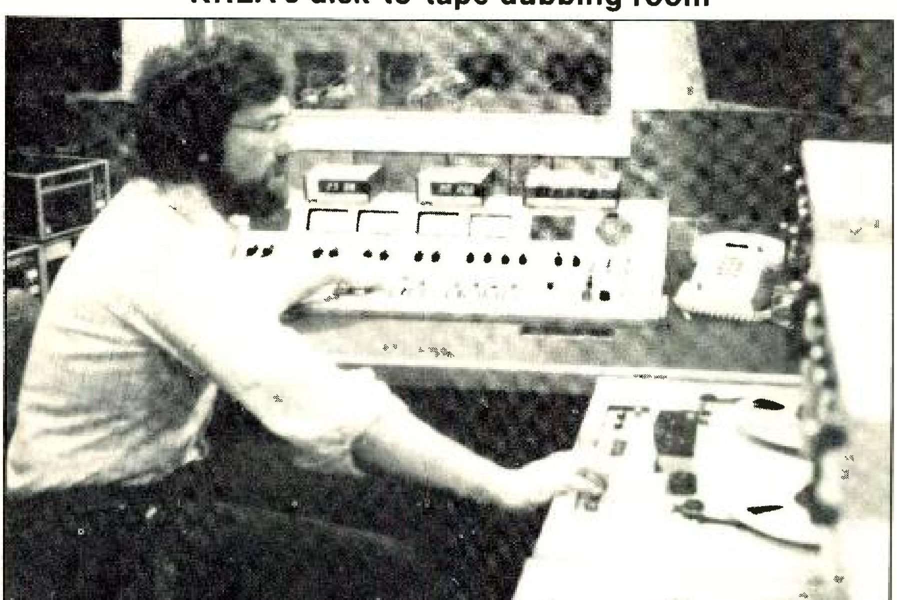
But other problems also can haunt those who cart their music.

"One station I was consulting," says Laird, "developed a terrible problem with their sound. It turns out that they ordered another batch of carts from the manufacturer, and the firm had changed the tape type, even though the station had specified the type in its order.

"What happens is you go out and wind your carts with one kind of tape, and when you come back to the manufacturer to buy carts with that tape on it, it's been discontinued, so you can't match the sound. There is not cart machine built that has selectable bias.

"With that one station, we were lucky. They had an extra record center, so we biased one for one type of tape, and the second for the other type, and recorded their music library that way. Once it's recorded correctly it will play back on any machines. But for a budget radio station, it makes for a real difficulty in consistency of the station's overall sound quality."

KRLA's disk-to-tape dubbing room



Machine alignment

Aside from tape selection, the mechanics of a cartridge machine also is an important consideration.

"There are trade-offs in the type of performance from cart to cart," Laird considers. "One cart will have really great azimuth control, and real bad wow and flutter problems; another one has super low wow and flutter, but doesn't have consistency in azimuth. If the cartridge has at least the same azimuth all the way through, you can adjust the record alignment so that it always plays back correctly, but if it changes on you halfway through, you're dead. We've compromised on what we think is reasonable wow and flutter and azimuth control with the Master Cart."

At KDAY, Laird has had his carts wound in lengths with 30-second intervals, to facilitate fast re-cue times without having to use fast-forward. "Fast forward usually creates azimuth problems that will muffle the first half-second of the song [when it plays again]. It leaves the tape pack loose at the point it stops, causing it to take that half-second to re-align properly."

"In the old days," Brown recalls, "you also got involved with record-head azimuth tweaking. You aligned the machine, set up the playback and record heads, and walked away from it. If a cart didn't sound good, you threw it away and started over. Using the old 300 Series Fidelipacs, the corner post is what determined the height of the tape as it came around the corner, and these things came loose all the time. Even the ones that were glued in firmly had subtly different heights, so every cart needed a different record head azimuth. Somebody had to stick an allen wrench into [the cart machine], feed a tone in, and tweak it for maximum output."

Azimuth requirements have become even more critical with the advent of the stereo cart, and FM broadcasting. "From the last figures I saw," says Brown, "about 63% of FM listeners listen to FM stations on table radios — that is, in mono — so it's *very* important to be top notch in left/right alignment to re-dub all of them, if you've put a house curve or some sort of noise reduction on them."

Level Matching

Laird recommends inserting all signal processing *after* the control booth, and separately processing the on-air personality voices and, if necessary, the commercials. This technique allows for greater flexibility in blending each of the elements into a cohesive station sound.

"Which brings up another point," Laird adds. "It's at the time of dubbing your music and transferring your commercials that the level decision should be made, because there is *no* way that a disk jockey is ever going to ride his level properly at the board. You can't expect him or her to fix something on the air that you had the time to fix while dubbing. I believe that all carting operations should have a set-up in the control room that puts the level control at the correct setting.

Technics SP10 MkII turntable and SME tone arm, mounted in isolation base

"For instance, here at KDAY the DJ just turns the pot wide open, and that's the correct level. It's a big help to the disk jockey, since he never has to think about how far to turn it up, and can never violate the console's headroom. If the cart has been dubbed right, and you fix it so he can never turn it up too high, it will *never* distort."

"When disk jockies are doing their shows," Brown offers, "they're thinking about something else; like, 'What's the next sequence? Am I following the format? I have to read this one-liner next, and then cue the newsman?' And all this. They are *not* thinking about the fidelity of the next record or cart, so the idea is to make it as *easy* as possible for the talent, or whoever is running the board."

Getting the on-air level right means that it has to be correct during the dubbing process. The available tools to do this are improving, but the majority of those involved in this discussion are still utilizing a VU meter, supplementing this broadcast standard with peak-reading lights on cart machines, and peak meters on reel-to-reel tape decks. TM Programming utilizes peak meters on the facility's Auditronics board.

"Peak meters are a good idea," says Laird, "particularly if they also reflect

No console. Nothing. Keep it as simple as possible."

Minimal Signal Processing

Although both TM and Schulke put their signal path through a fader on a control board, they also support the basic philosophy of "less is more." In particular, all those interviewed on this subject were particularly cautious about the use of equalization.

"When we transfer a record for the first time to build a show," Joel says," there is *no* EQ added — it is transferred *flat*. When we put the show together then, in the context of the surrounding pieces of music, we decide to go back and pre-record a master with some very gentle equalization because, the first record was made, say, by RCA in 1959, and the second record by Capitol in 1980. There has to be some adjustment made in order to make those two pieces of music compatible with each other."

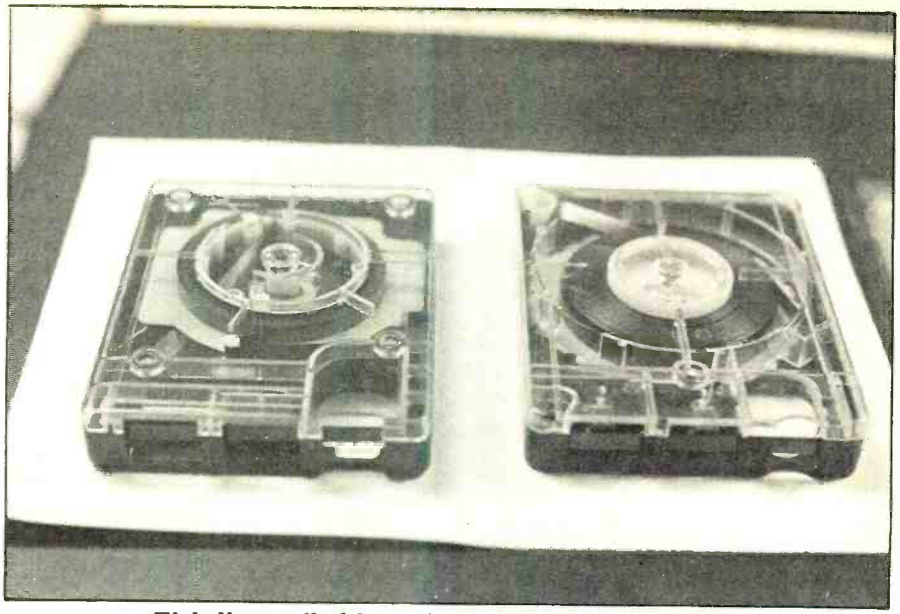
In an AOR or country or rock and roll situation, those sound differences can be important to retain in the program.

"We really try to stay away from equalization," says Hogan. "Our philosophy is *flat* response. When the station gets a hold of our tapes, they are going to run them through their broadcast chain. If we've already compressed and limited and equalized, and then they run it through their chain, and compress and limit and EQ the material, what you get is a very *dull*, fatiguing sound.

"Some people use what we call a 'blanket EQ': they want every song, equalization-wise, to sound the *same*. They want that nice, smooth, consistent sound. Some people feel that it will make people listen a little longer. We don't agree."

Programming considerations also apply, particularly with the older records. While occasional adjustments have been made on extremely poor recordings, first broadcast in the days of less sophisticated home equipment, Brown holds to a minimalist theory of EQ: "At KRLA we deal in memories, and we want [the song] to sound like they sounded originally. Some of the older records present problems, if they have an irritating mid-range, or no low-end at all. On those records we try to give the bottom a little more of a chance, but the whole idea is to make it sound as *natural* as possible. We're *not* trying to make a record recorded in 1962 sound like it was recorded in 1970.

KRLA was fortunate in this particular aspect of transferring records to tape, Brown considers. "Dave Hull, our afternoon drive time personality, was very closely identified with the Beatles here in Los Angeles. He and his wife recently came into the station with a virgin set of the English-release Beatle albums, still in their shrink wrappers. I just reloaded all those pressings into our library. The records are *very* clean and quiet, and they're subtly different from the US mixes in some cases. What Capitol did to the albums before they released them here is not very subtle; sometimes they really messed with the EQ and added reverb, or used alternate takes."



Fidelipac (left) and 3M Mastercart NAB Cartridges favored by KDAY's Andy Laird

KDAY's Tape Dubbing and Multitrack Production Facility

Use of compression also is kept to a minimum by all those involved in this discussion. "On classical music," says Hogan, "it's *very* noticeable. Classical listeners are usually audio purists; they know when a passage is being kicked up, or when it's going down. On beautiful music, you have to ride some gain. There are so many low, low passages, and overall it is best for us to bring these up from the disk. True, signal-to-noise suffers some, but not like it does when the station's compression goes 'looking for level' down in the tape hiss, and all the stages after the tape hits the client's tape heads."

"There are dynamic ranges on certain records that are beyond broadcast capability at this time," says Irv Joel. "So those are supported gently, but manually. Say you look at the average [level] of the whole record, and for the entire 3½ minutes it's 'X,' but on two places the tympany hit 4 dB over. Rather than cut the whole [track] at 4 dB under to make sure those tympany hits don't overmodulate, we would gently level ride those hits. Compression, or any automatic [gain-reduction] device, is going to undo the dynamic range — make loud passages soft and soft passages lound — and that's not what we're into at all."

There are other practical reasons for keeping the transfers as flat as possible, particularly with disk-to-cartridge dubbing. "You want your library carted as near flat as possible," says Laird. "If you record 1,000 carts, and then you want to change the sound of your radio station, now you've got to go back and

Pat Hogan/TM Programming

back a turn. Then I'll start the turntable, and [when the stylus reaches a set mark] I'll start the cart machine. That way it's truly up to speed, and any mechanical vibrations will have dissipated by then."

Schulke and TM, both of which favor sub-mastering their material on to reel-to-reel tapes, avoid back cueing all together. "We do not back cue," says Joel. "Everything is layed down straight away. I was with Capitol records for 15 years, and did cutting and mastering in New York City. I'm familiar with this kind of thing, and if you're careful — and we are *very* careful — you can play a record a hundred times without any terrible degredation."

The moving coil cartridge also takes considerations further down the transfer chain, as Laird explains: "It has an extremely low output voltage, and you have to use either transformers or a pre-amp in front of the phono pre-amp. I think the best combination I have heard has been using moving coil cartridges with transformers made by Deane Jensen. The phono cartridge leads plug into the transformers, and the transformer leads plug into the phono pre-amp."

For Laird, pre-amps are a particularly important aspect of the audio chain. He has long preferred to use consumer phono pre-amps for their superior fidelity. "It hasn't been until recently that there have been some industrial oriented turntable pre-amps that have really good sound quality," he offers.

Irv Joel at Schulke holds a similar point of view. "A word you hear a lot here," he says, "is 'modified.' Not that we like to do it, but it's the only way to achieve what we need. The pre-amp we're using at the moment is made by Audio Interface, and we modify it once it gets here. We do not use transformers where they are not necessary, and those that we do use are the high quality Jensen transformers."

TM also uses the Audio Interface pre-amps. "They're hand wired," Hogan adds. "A beautiful piece of equipment; transformerless with internal Mu-metal for RF shielding and separation. Our engineer will set the curve in each pre-amp for that particular room, according to what the [acoustics] requires."

"I also use the Audio Interface," Laird offers, "which is very expensive — we're talking $1,000 — but that's for the one turntable on which you're dubbing music. Everything that's coming off the air will [be dubbed] off that turntable, so the investment is worth it — especially when you consider you might spend a couple of grand on a decent broadcast turntable, and how many minutes an hour would that play. I have spent nearly $3,000 on one turntable set-up: cartridge, tone arm, table and base."

There are other phono pre-amps that have impressed Laird, including units from RTS, Straight Wire Audio, and Broadcast Electronics, but he has had good experiences with using a consumer version to interface with a more reasonably priced dubbing-room installation.

"Starting in about 1976," he recalls, "I was using the Onkyo P303 for radio station installations. It's easy to interface into a commercial situation and, by just building a VU meter amplifier for it, it has enough drive to input directly into the cart machine. That is part of the goal — to keep the total amount of electronics involved as small as possible.

"This is another rule for dubbing music: dub your music with the *least* amount of electronics possible — straight off the turntable, through the turntable pre-amp, and, if you can, directly into the cart machine. *No* EQ.

Douglas Brown at KRLA has gone so far as to have collectors bring in rare records for dubbing on to 15 IPS tape, so that the station forever will have a copy. The condition of records is also a concern in the transfer process: "I use the Disc-Washer," Brown says. "That's a particularly good system for removing static. We have a lot of records here that the disk jockies actually played in the Sixties. Sometimes you can take a record that you thought was a lost cause, and make it sound real good with the Disc-Washer."

Schulke and TM favor a more elaborate record cleaning system. "We have a Keith Monks Record Cleaner," says TM's Hogan. "You lay your record down on the platter, hit the switch to 'wet,' lay the camel hair brush across the record, and hit a pumper button. A solution will be spread on the record evenly by the brush as it cleans all the gook off the [rotating] record. Once this is done, you kick it into 'dry,' a suction motor come on, you put what is basically a tone arm on the end of the record, and it tracks with a small thread in the groove, digging out anything in there. At the same time it sucks this and the excess solution off the record. There is nothing that can match it."

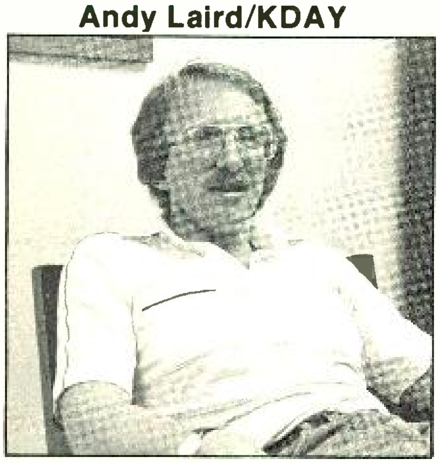
Disk-to-Tape Transfer

With the best possible disk selected for programming, the transfer process begins.

"With many of the stations I work for," says Andy Laird, "we will devote a room for dubbing music on to cart. It's a room specifically devoted to controlling the air sound, and has a really good monitoring system that can also be switched to monitor the output of the turntable pre-amp, cart machine, broadcast booth, each audio processor, transmitter, and finally over-air. You can listen to each stage of the chain, and get a good handle on improving your sound.

"Another point," he continues, "is to have *one* person do the dubbing to tape. If you have five or six different guys dubbing your music, you'll have five or six different versions of what the right levels are, and how things should be done."

Andy Laird/KDAY



Douglas Brown/KRLA

"It really pays to have one set of ears doing it," agrees Brown. Otherwise, "you're going to have varying degrees of looseness. Cues are going to be in different places on the carts, and it won't be as tight."

Regarding the type of equipment necessary for disk tranfer, Laird has even gone into the consumer electronics area while outfitting the stations he services. "Initially, broadcast turntables had rotten noise problems; they sacrificed rumble for rapid start. Now there are a couple of super quiet, fast-starting turntables available that are very expensive. While it is absolutely necessary for on-air work, rapid start is *not* needed for dubbing music, so I'd go for a good-quality, direct-drive consumer turntable with low wow and flutter and rumble that will take a really state-of-the-art tone-arm."

Isolation of the turntable depends on the studio location, and the stability of the building, but still two different approaches became evident in this discussion. Both Hogan at TM, and Brown at KRLA, have ensured isolation by utilizing independent turntable pedestals.

"They're designed by Pacific Recorders in San Diego," says Brown, "and there's about 45 pounds of sand in each one of these four-spring loaded guys; it does a very good job of insulating the table from both shocks on the floor and [acoustic] feedback. If you bash a cabinet it's going to move, but we're on the second floor here, and can't put a concrete pillar into the center of the earth."

Laird and Joel at Schulke, on the other hand, have taken a less involved approach in their installations.

"I think that most consumer turntables, and the base that is sold with the Technics SP-10 MkII, have *incredibly* good isolation from the surface where they sit," Laird considers. "Lately, I've been building a turntable 'commode' where the turntables sit, as opposed to building a sand-filled box into which they mount. In fact, the ony turntable base I recommend is the optional onyx base sold with the SP-10 MkII. It's very expensive, but *absolutely* isolates the turntable from the table top on which it sits. It also keeps the turntable from making any noise when it starts."

At KRLA's production facility in Los Angeles, Technics SP-15 turntables are fitted with Audio Technica tonearms and Stanton 680EE cartridges, a tonearm/cartridge configuration that is adaptable to varying situations.

"The tone arms are very accurate," Brown says, "and Stanton cartridges have a very smooth frequency response, and are pretty durable. They're elliptical, but sometimes we will go back to an old 1-mil conical stylus to track an old single that was cut fat."

At Schulke, Irv Joel's Technics turntables utilize an SME tone arm, as does Pat Hogan at TM. "Our turntables are modified to handle SME Model 3009 Series III tone arms," says Hogan "The tone arm itself is super light, the base of which is balanced in all three axes, and viscous damped. For reasons having to do with resonance, our engineer has put acoustical putty between the headshell and the cartridge. We use Stanton 881S cartridges, which we find are a bright and durable studio-reference cartridge."

Simple Signal Paths

Andy Laird has been experimenting with a more consumer oriented configuration: "I have had a good deal of luck with moving-coil-type cartridges and SME tone arms. The really fine detail in records seems to become audible with a really good moving-coil cartridge. This mid-range articulation lands right in the range of what these inexpansive, high-tech home sound systems of today can reproduce. I use the Ortofon MC20; other really great sounding moving-coil cartridges are Denon, and Fidelity Research.

"If you're using a broadcast cartridge that is built for control-room use, they may have a maximum tracking weight of 3 or 4 grams, and you're asking for destruction, especially on 45s. On a moving coil, I'll run the phono cartridge up near maximum tracking weight, but that maximum weight is only 1.5 grams."

Also, in the transfer to tape process, emphasis is not placed on rapid start. "I take a full-turn," says Brown. "For an LP, I'll use these markings on the turntable. I'll cue it to the first audio, and go

Irv Joel/Schulke Productions

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DISK-TO-TAPE TRANSFER

Maintaining Quality Control During This Critical Broadcast Production Process

by Steven Barnett

Since so many radio stations in the past years have been programming from tape rather than disk, it is timely that *APfB* should examine the ways in which music or production audio reaches that medium. The transfer of a record cut to tape should be a carefully considered process, as carefully considered as the decision to format on tape, be it cart or reel-to-reel.

Distribution of programming material on reel-to-reel tapes, as practised by Schulke, TM Programming, and many other radio syndicators, has been prevalent for a number of years. Music programming on cartridge at automated stations also has enjoyed a similar industry-wide acceptance, but now an increasing number of live radio operations are switching to the use of carted music, rather than disks.

Andy Laird is chief engineer at KDAY, Los Angeles, and serves as an engineering consultant for 30 radio stations nationwide. In his advice to client stations, Laird places great emphasis on transferring program material to NAB cartridge. "With carts," he offers, "you can maintain better overall sound quality on your music on a day-to-day basis, than you can by playing records. If you repeat records with any frequency, you're subjecting that record to heavy groove fatigue. Disk jockies are *not* the world's best record handlers; they have to move fast, so the record goes through all kinds of mishandling. It's grabbed by the grooves, not cleaned properly, and all the rest."

Carting provides a station with control over quality, says **Douglas Brown**, production director at Los Angeles' KRLA. "Disk jockies are notoriously hard on records. That's coupled with the lack of quality in records . . . they're using recycled vinyl, which is very soft, so that one back cue and there's a cue burn on the front of the song."

"Additionally," continues Laird, "because of the nature of broadcast studio use, you can't utilize the best phono cartridges and tone arms in the pickup system. This problem *alone* makes it worthwhile carting music. Phonograph cartridge and tone-arm combinations that are rugged enough [to be reliable] in everyday control rooms provide an inferior sound quality that is audible on most radios."

Quality Source Material

To start the transfer process, one must find the best possible source of material, which is not always limited to disks. **Irv Joel** is the director of field engineering for Schulke Productions, New Jersey, radio syndicators of beautiful music format programming. Schulke, Joel considers, has certain advantages as far as access to source material.

"About one-third of what we are producing now is exclusively custom to us," he explains. "It is music that we have recorded for us in the United States and Great Britain, and is received on two-track tapes."

A large part of Schulke's program material is still sourced from disk, however. "In 12 years in business, we have amassed a library which is really priceless," Joel continues, "because we just can't get some of this material anymore. It's out of print completely, but Jim Schulke and our music director, Phil Stout, were smart enough to have acquired multiple copies of these very rare records. The disk, believe it or not, is a *very* stable item — more stable than tape. We keep them in a vault at room temperature, and watch out for dust and that kind of thing."

This mastering process is sometimes for storage, but in Schulke's programming the direct-from-disk tape masters are assembled into the master tape for a program, which is then duplicated for release. Radio stations with other formats, however, also have a tape option.

When Douglas Brown was working at Los Angeles' KHJ in 1976, he says, "we would get a lot of singles that were edits of album cuts that sounded awful, so we asked if we could get a reel-to-reel of the singles. We sold [the record companies] on the idea that everytime we play their record, we want it to sound the best possible — no 'pops' and 'clicks,' and distortion. Now, we get the current records on 15 IPS stereo tapes from the labels."

Some record companies tried to get away with sending their own dupe off the record, but Brown was alerted by the fact that uninformed engineers would leave an audible needle drop on the tape. For older singles, where tapes from the stereo masters are unavailable, he prefers albums to singles.

"We try to use new LPs whenever possible," he adds. "Now in many cases — the US Rolling Stones' material until 1966, and *Aftermath*, for instance — the albums were cut in mono, and now are only available in 'reprocessed' stereo. Psycho-acoustically, it's *not* right, and if you've ever heard these pressings on a car stereo, it sounds like heck.

"We'll load these things in mono, but on some [record company] reprocessing techniques they add reverb. So, when you combine the left and right — which theoretically should sum up to the original mono, EQ and all — it doesn't work. In that case, we go back to the single, and find the cleanest copy we have. Since KRLA goes way back in playing pop music, we do have a lot of that [older material], and some of it is in reasonable shape."

TM Programming faces these same problems in the production of its seven program formats, ranging from beautiful music to country, to album rock. **Pat Hogan** is the operations manager and production director for the Dallas-based firm: "We do get pretty good service from the record companies, but a lot of times we'll just have to go out and buy the record, especially the old stuff. I have a head librarian who is a research wizard and a record freak, and he's just a *bloodhound* for old records."

All the music at TM Programming is sub-mastered on to tape for storage, and ease in assembling the program material. In transferring from the original disks, problem records often call for special treatment. "We go back and manually 'de-pop' the record on the sub-mastered tape," says Hogan, "by scraping the oxides on the tape, manual splicing, and what have you. We have a couple of devices that we use for high-frequency surface noise, but we like to stay away from any processing or EQ at all.

— *continued overleaf . . .*

talk station studios serving as visual intercom systems.

Because the microphone is live virtually all the time in news and talk stations, it has been difficult for producers to communicate with air hosts during a program. Air hosts and programmers for years have bemoaned the fact that producers often exercised more control over call-in talk programs than the air personality. Producers in one booth have a wealth of information about callers standing by to go on the air, while the host in another booth is flying blind, often unaware of the number or quality of calls waiting. In a number of stations, computers are now used to display caller information for the host, permitting greater control of the program by the latter.

One of the first to utilize a computer for the purpose was David Graves, general manager of WIND, Chicago. Graves first introduced a computerized display system in 1978, when he was WIND's program director. He created a program for his personal Apple II home computer, and brought the system into the studio. Within three days, Graves says, the system became a necessity, and he was about to loose the use of his own computer. Today the station owns its own system, complete with programs that have been revised and improved by Graves over the years. The system is a single Apple II in the producer's booth, with slave video displays in the on-air talk studio, and adjacent news booth.

The current area weather forecast is constantly displayed across the top of the screen, eliminating, as Graves says, "the need to carry one more piece of paper in an out of the studio." In the area of the screen below the forecast the producer can key in special messages to the host, including news bulletins. Bulletin encoding can include details of a breaking story, or instructions for the air host to "switch to the newsroom." Such urgent messages flash on and off the screen before the air host.

The lower portion of the screen displays the status of calls holding to go to air. The call information includes the first name of each caller, the topic they wish to discuss, and their point of view — pro or con — on issues being aired. Thus the host can orchestrate the flow of calls to keep the program moving briskly along. Graves says the cost of the Apple II system and peripherals was $2,300, and points out that a standard electronics studio audio intercom can run as high as $1,000; hence, the premium for the constant communication provided between producer and host is little more than $1,000.

WIND producers quickly caught on to the computer language, Graves says, and the simplified program made it possible to train incoming producers in a few hours.

Other stations have elaborated on the system. KSTP in Minneapolis, using a Radio Shack TRS-80 computer, inputs the age of each caller into the system, so that the demographic profile of callers can be monitored on a daily or weekly basis. WBZ in Boston uses its computer to store data on contest winners. A heavy user of contests, station producers put winners' names and addresses on the computer, and recall them the following day for production of mailing labels for prizes.

Call-in Programming

Computer technology also is being applied to the enhancement of telephone audio quality. KSTP's Mark Durenberger points out that micro-chips being developed for use in data transmission will have applications in Talk Radio. The chips can be adapted for use in eliminating the hybrid leakage of broadcast phone circuits — the cause of the "hollow" sound often heard on many stations when a phone line is on the air. Furthermore, Durenberger says, once the hybrid leakage has been eliminated, processing can be applied to the telephone signal using techniques now common in recording studios, to reshape the sound of the telephone caller's voice with noise gates and parametric equalizers. While the chips that form the heart of the system are still on the drawing boards, Durenberger anticipates the systems could be in use in talk studios in a year.

With the Talk Radio format proliferating on the AM band across the country, we will certainly be seeing more and more stations innovating with systems and equipment to bring down the cost, and increase the efficiency of information production. ■■■

WMCA talk host Barry Gray
at a semi-automated console

TECHNOLOGY FOR TALK RADIO

Improving the Production of Talk Radio Programming for AM Broadcasters

by Bruce Marr

At no time since the advent of television in the Fifties has there been such a state of turmoil in commercial radio broadcasting in the United States. With FM now the dominant medium in most of the nation's cities, AM operators find themselves seeking new and different kinds of programming that will again attract listeners to their side of the dial. And because most broadcasters concede that FM is becoming recognized as *the* medium for music among the stereo generation, a fair number of AM broadcasters are turning to news, talk and information formats in their efforts to compete.

While the news and information format offers an alternative for many AM stations, there are some inherent drawbacks to face when considering such programming. Talk programming requires that a considerable number of extra man-hours be devoted to programming by producers, researchers, call screeners, and guest bookers. The format demands on-air hosts who are willing to prepare hours of material for each day's program, and are articulate when they reach the studio. As a result, most broadcasters find information programming to be significantly more expensive than music formats, which require only a disk jockey and a stack of records to sustain a three- or four-hour program.

As more and more stations turn to the Talk Radio format, programmers and engineers are developing new systems and techniques to enhance the quality of their programs, and minimize the costs. Often the innovations utilize new and existing technology, and computers.

Historically, talk programs on major market stations have required at least three people to get the show on the air: the on-air host, an engineer, and a producer or call screener. That's a substantial increase in operating costs over a disk jockey program — significant enough to turn many broadcasters away from the format. Many stations found an engineer to be necessary for a talk program, because the host was on the air before an open microphone virtually every minute during the program. There were no records to provide time for the host to load cart machines, and take care of other engineering chores. Talk hosts could not be expected to be at their best while interviewing a guest or talking with a caller, if half of their mind had to be devoted to engineering the program.

An interview program can only be as good as the interviewer's ability to concentrate on the subject being discussed. The host discussing, say, nuclear energy with a guest must be free to draw on his own store of knowledge of the subject, perhaps refer to written material and notes, listen to the answers of the guest, and at the same time formulate new questions. This is difficult enough under the best of circumstances, and much tougher when the host must also superimpose the necessary technical and production chores.

Similarly, because most stations were not willing to permit listeners' calls to be aired without prior screening, the producer or call screener was required to answer the incoming lines and determine the quality of each call, before turning it over to the host.

Custom-designed Consoles

Seeking a way to permit its air hosts to operate combo, WMCA in New York developed specifications for a talk studio console that minimizes the attention required of a talk host. Using the specs developed by the WMCA engineerring department, a console was custom built by Logitak Electronic Systems, Inc., of Houston, Texas. Among other features, the system controls all signal levels automatically. Using noise gates and modified compressors, the console optimizes the level of all microphones, telephone inputs, cartridge machines, and incoming network lines. While there are VU meters on the console, theoretically they could be done away with altogether.

The switching required of the host has been greatly simplified. While the host turns on any of the four studio microphones as required for himself and in-studio guests, all the microphones automatically are muted when the host triggers a cartridge machine, or the network feed. All cartridges are preloaded into nine machines, and then triggered by a single button in the selected order.

Even though many of the talk show hosts at WMCA have been on the air at that station for many years, they have never been required to acquire technical skills until the new equipment was installed. Nonetheless, the engineering staff reports that all of them quickly adapted to the systems, and the on-air sound has been enhanced.

Computer Applications

Just as the computer is finding its way into our personal and business lives, microcomputers are turning up in soundtracks. What is new, however, is their application to the home screen. And, according to Werner, that application will have an effect on the movie industry, too. "As television gets better, and robs audiences away from movie theaters, filmmakers will hae to do what can't be done on TV: surround sound, giant sub-woofers in the back. If television is then going to keep up, it will have to offer something besides just 35 cable channels of pictures."

Preparing for the Future

Werner sees television today undergoing a revolution similar to the one that changed the face of home audio in the Sixties. "Before that," he says, "your 'Close-N-Play' phonograph was about it. It was good enough to dance to, but it wasn't what anyone would call 'high-fidelity.' It took a little while to grow into that, and then it exploded." To take advantage of the next explosion, Werner concludes, television producers and sound engineers will have to adjust just as much.

"We have to bring television sound up to the disk standards of the last five years," he says. "For a long time, sound in video was treated as a necessary evil. Generally, very little was done to prepare for an audio session — if the VU meters were kicking there was sound, and that was about it. The room may have had peaks and dips of 5 dB or more [in its frequency response], but that didn't bother anyone. Now we have to do a little better — there's a basic attitude change that's necessary."

Werner lists several ways, large and small, in which video production studios can improve their audio. Generally, these techniques involve simply adhering to audio-studio standards: simple things like laying Dolby noise-reduction levels, azimuth, and frequency-response tones on the audio tracks of videotape reels, *not* just 1 kHz at 0 VU. Another thing, which he says "people hesitate to do," is aligning the audio sections of VTRs when they are installed, and periodically thereafter — "not just running them right out of the box." Then there are the niggling little chores that audio recording studios always have to stay on top of: making sure all lines are balanced and phased properly, and keeping electrical noise and RF interference out of signal paths. Finally, antiquated equipment has to be replaced with modern, professional-quality consoles, tape decks, noise reduction, and signal processors.

It is just as important, he feels, to bring new talent to the fore, both in front of the camera and behind it.

"There's been a tremendous explosion in TV outlets," Werner says. "There's been no problem with money for developing them, but personally I think there have been problems with money for new product. Lots of small studios with limited video facilities are trying to be creative, but they can't get [their output] on the air. The way I see it, all this new technology is still being used in traditional ways in all the old places — New York, Los Angeles, Chicago — and a lot of the executives buying product for the cable services are the same people who used to do it for the networks.

"It's also weird that I can't get a *single* stereo audio track. MTV, whether it's good or bad, has at least made that an issue, whereas a year ago, nobody cared about it at all."

As for finding the right people to man the frontiers of audio for video, Michael Werner is very optimistic. "Because of the unfortunate things that have been happening with the record business, there are lots of well-trained, talented audio types walking around. That's fortunate for TV — we can assimilate them and teach them a little video, which will do us all good. There'll *always* be work for good people."

Investing in the right people and the right equipment to upgrade the quality of television sound is only half the battle, however. "Better sound means more *time*, and that means more *money* from the client," Werner concedes. "Production houses will have to show clients that audio is important enough that people wil prefer their product to someone else's."

When the program sponsors and providers, and the television receivers manufacturers, finally opt for quality sound in a big way, Werner feels that they will be rewarded. "The payoff will be that the audience will hear the difference — and they'll *listen*." ■■■

behind the picture, as it may after many tape passes, because of the time difference between the replay (old mix) and record heads (new mix data), it can be bumped forward by using the sub-frame lock capabilities of the Shadow system.

The final mix was done to digital two-track through a Sony PCM-F1 16-bit unit, the output of which was recorded on a separate ¾-inch U-Matic videocassette, striped with timecode. "You can't put digital on the air," notes Werner, "unless you get a second video transponder for it on a satellite — although someday, with Direct Broadcast Satellites, it'll be easy. We use the digital tape to keep down generation noise. Because it has timecode on it, we could even use it, if we wanted, to maintain master quality right up to the feed of a cable network or system — that is, if the uplink or the head end has another F1."

Digital mastering is not the only technique Broadway Video uses for preserving sound quality. Another method, used on some of the Simon & Garfunkel mixes, and one that Werner would like to see more clients take advantage of, is to bypass the two-track tape stage completely, and mix directly from the multitrack to the audio tracks on the video tape. "That's about as good a sound as it can be," he says.

Vision mixing desk (left) and sound board on rear level enable close communication between engineering staff.

Mixing for Video

Does mixing music and sound for video pose different problems from mixing for records or radio? Michael Werner feels strongly that it does. "The difference is not that great with regards to equalization or limiting," he argues, "although there are things to watch out for. On videotape, you have to make sure that the audio isn't so strong that it starts to modulate the picture — with herringbone patterns or whatever — and vice versa. You also have to keep in mind that most home TV sets can't handle a lot of low-end, so sometimes we'll roll off 18 dB per octave below 80 Hz for a TV-only mix. And when you're going over the air, you have to worry about FCC regulations.

"Where the main differences lie are in the way the mix *moves*. Most live concerts over TV today don't move at all —the audio doesn't match the action. That's because the [engineers and producers] doing them are using the old criteria of records: the audience sits at home with no visuals, and recreates the scene in their heads. In that case, all the mix has to do is keep the individual instruments distinct and clear. With a picture, however, people *see* movement, and they expect to *hear* movement to match. There are lots of things that have to be conveyed on a video soundtrack that are not germane to record producing. For example, if a guy trounces a guitar across a stage, you should pan the guitar a little — if you're doing a stereo simulcast or track. If he walks down a runway or towards the camera, you bring him out of the background a little. If the camera is tight on the drummer, you want to hear more drums.

"Nobody really knows exactly how to do this," he laughs. "We just try, and sometimes it works, sometimes it doesn't." All of the capabilities of the mixing console and the processing gear — pitch shifters, compressors, reverb, and equalizers — are explored. "It might be a rougher mix than some people like to hear," Werner admits, "but I think it's more important to recreate the feeling of the live show, than to do a record-perfect mix. Naturally, the live audience doesn't hear it the way we mix it, but they don't have to — they have their own input, because they can focus their eyes on whatever they want, and their brain will pick out the sound. They also get more of the hall sound than we would put in a mix, because they can distinguish between it and the sound coming from the stage, which isn't as easy at home.

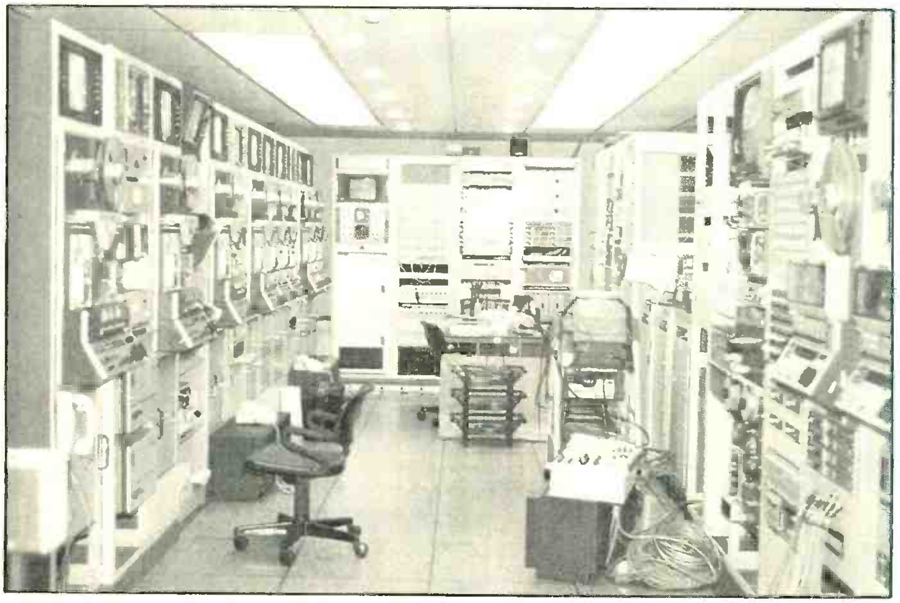
"By switching the video, we're giving the audience cues as to what to watch —we've sorted it out for them in post-production. But just having the visual cues and a straight audio track isn't enough. The audio has to be subtly altered so people get the feeling that they are there, that they are being dragged along with it. It has to convey the energy of the event."

The most important parameter in this kind of mix is, of course, the relative levels of each element. But other tricks can be used as well. "We use time shift, to indicate distance and movement," says Werner, "and reverb, panning, and gating, in addition to changing balances.

"It can easily be taken to extremes," he warns. "It's got to be *subtle*, not vulgar. It puts a lot of responsibility on the director, and the audio mixer. They have to talk to each other."

Anyone familiar with film sound will recognize that these techniques are not new — they've been used in motion pictures for years, and Broadway Video, not surprisingly, does its share of film

Video tape machine room equipped with Sony BVH-1100A and -2000 decks.



of these media make different demands on the audio production department.

Supervising the broadcast sound was well-known audio producer Phil Ramone, while long-time S&G engineer and producer Roy Halee handled the audio for the record release. A Record Plant Mobile recorded the concert on 24-track tape with timecode, and then Broadway Video took over.

"First, we pared down some of the intro charter and speeches," Werner explains. "One musician's intro was covered by crowd noise, so we overdubbed that. We also redid one of Paul Simon's guitar tracks, because his microphone went flaky at one point. Phil's stereo mix was done at Soundmixers [recording studio, New York], and we went back and forth with two-track tapes between their room and ours, listening on several pairs of monitors, until it was right. Halee's album mix —also done at Soundmixers — was more spacious and 'open,' while Phil mixed it for broadcast somewhat closer and more punchy — a little less reverb, and more compression.

"One of the things we had to do was negate the PA effects; because the crowd was so huge, there were delay lines on the sound system, and we picked up a lot of slapback, particularly on the crowd mikes. We ended up lending soundmixers our MTR-90 and Shadow, and put the music on to 16-track — without the crowd sound. We took that tape and mixed it back on to two pieces of one-inch video, and put the crowd on a separate piece of videotape, all with timecode. Another reel of video had the final program cutaways. When we assembled it, we 'checkerboarded' the songs on the two tapes, and bounced back and forth between them, using the crowd tape for segues.

"Sometimes we pushed the main sound back; sometimes we brought the audience forward a little. Halee didn't have this problem on the album, but for the videotape, we had to make sure that the audio was always in sync with the picture.

"For the mono [satellite] mix, we used a composite of the stereo. We borrowed an old mono box from CBS that they used to mix stereo masters to 45s, which rolls off below 100 Hz, eliminating the bass build-up you get otherwise."

The company also was involved with producing a program shot during Neil Young's recent European tour, which it has sold to HBO. This assignment required even more prestidigitation.

"We hired crews in Europe for three dates in Germany," Werner recalls, "and sent over our own directors and producers. Audio was recorded on 24-track, but the timecode was [50 Hz; 25 FPS] EBU, not [60 Hz] SMPTE, and the video was shot in PAL, with 625 lines and 50 fields per second. Image Transform in California converted the video for us to NTSC 525/60, using their ACE four-field [standards conversion] system, but we had to hand-sync the timecode on to the audio tape.

"We established the proper offsets with our Sony editor and Shadow, and synthesized a 50-Hz capstan drive signal, which 'faked' the Otari and the Shadow into thinking they were back in Europe, so that we could restripe the 24-track with SMPTE code. We kept the PAL around for a while on the tape in case we had to do it again, and also put down a 59.94-Hz [drop frame] resolve track.

"We had to do some fixing on the audio, too. We weren't happy with the live room sound, so we redid that. Also, the percussion tracks were weak, so we overdubbed them here, as well as some of the synthesizer and vocoder tracks."

Singer Neil Young supervised the mixdown himself, some of which was done at his ranch, in studios in California, at New York's Power Station Studio, and even on the road during the rest of the tour.

The secret to keeping the mixes straight was in the mixing console automation system, says Werner. "Sure, there will be differences between the various studios even if they use the same automation equipment" — which, obviously, they all had to do to ensure systems compatibility — "but they shouldn't be any greater than the differences in the monitoring environments themselves." When the automation lags

— *continued overleaf . . .*

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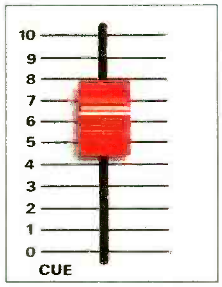

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INTERFACING "SEMI-PRO" AUDIO PRODUCTION AND PROCESSING EQUIPMENT TO THE BROADCAST PRODUCTION CHAIN

by Lou Schneider, Director of Engineering, Visionary Radio Euphonics, Santa Rosa, CA

Studio equipment is generally catagorized as being either "professional" (pro), or "semi-professional" (semi-pro). Sometimes this description defines the ruggedness — and price — of a piece of equipment. But, more importantly, from the installer's point of view it is a classification of the input and output circuit electrical characteristics, and interface capability.

As it is well known by most production engineers, the two main catagories are either "balanced" or "unbalanced" circuits. An unbalanced circuit is the simplest, and utilizes a single conductor for the signal lead, surrounded by a shielding conductor. This outer conductor serves triple duty: signal return; system ground; and interference screen. The interference to be screened is generally from outside sources such as radio transmitters, lighting, power-line radiation, and so on.

A balanced circuit uses two conductors for signal transmission, the outer conductor serving *single* duty in screening outside interference, The audio signal is now isolated from the system ground, the latter being obtained by bonding each piece of equipment (chassis) together in a rack, or via the power line (third prong) ground return — not always recommended. The audio cable shield is connected at only one end, and this can be either the sending or receiving piece; it should be consistent, however, throughout the installation. In some cases the shield may be connected at both ends to provide or "improve" system ground, but this procedure should not always be considered normal practice, and is usually an indication of grounding problems.

Balanced circuits provide greater immunity to interference signals and ground loops. Since current in each conductor flows in the opposite direction from one another on the signal leads, any induced hum noise will cause a curent flow in the *same* direction on both signal conductors. This interference will be cancelled out by the balanced circuit — an effect referred to as "common mode rejection," and a very important consideration of balanced lines. Since the cable shield is not part of the signal path, one end can be left "floating" to break up ground loops.

An unbalanced circuit, on the other hand, does not afford such protection from interference. If both ends of the return lead/shield are not at exactly the *same* potential, any signal induced upon the shield will ride with the audio and be amplified in the input stage along with the desired signal.

Mixed Balanced and Unbalanced Working in Production Studios

Not all studios need to be operated as strictly balanced or unbalanced. Unless you are located in a high RF field, unbalanced operation will deliver very good sound — provided, that is, the limitations are understood, and the potential traps avoided. The major difficulty in working with unbalanced circuits will be proper grounding, and the interface of semi-pro and pro equipment, since the two systems quite often operateatdifferentaudiolevelsandinput and output impedances.

Pro equipment input and output circuits operate in a balanced configuration, with an output impedance of 600 ohms, and either a 600-ohm input or 10k-ohm "bridging" input. Standard operating levels in and out range from 0 dBm to +8 dBm. Semi-pro equipment, however, operates unbalanced, with typical input and output impedances of 10k-ohms. Operating levels are normally 20 dB below that of pro equipment (typically -10 dBV for TEAC Tascam and

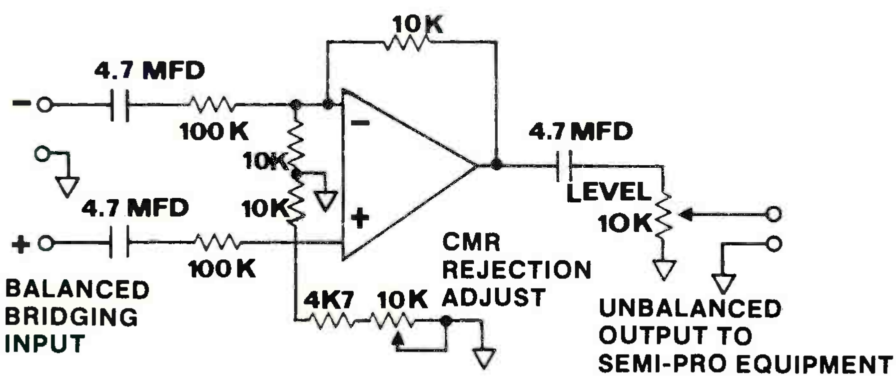

FIGURE 1

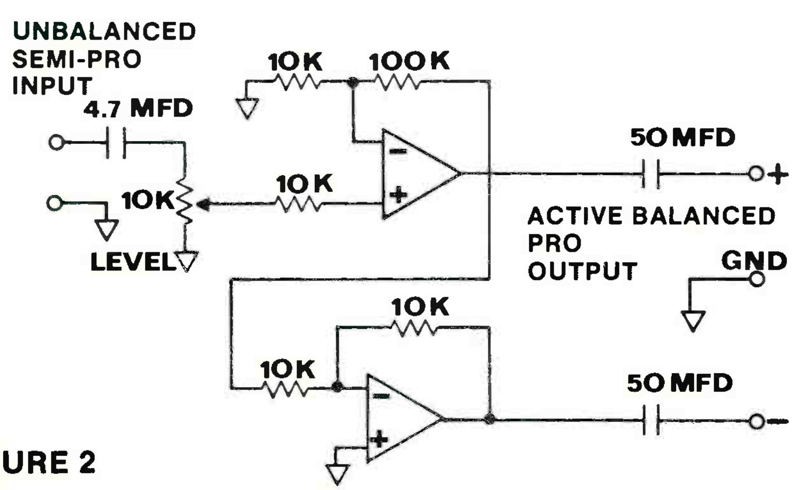

FIGURE 2

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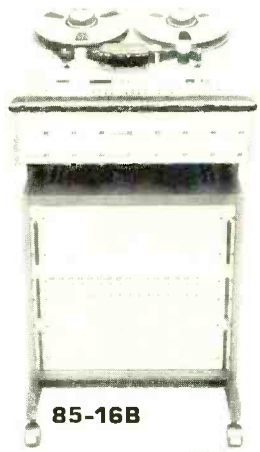
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85-16B

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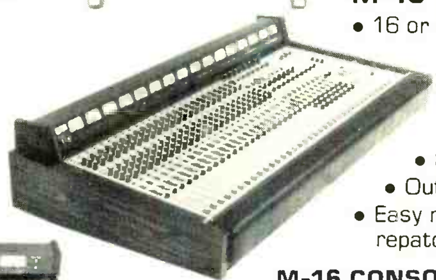
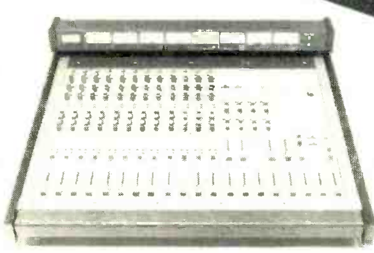

M-16 CONSOLE

M-16 Console

- 16 or 24 inputs, 16 meters
- 8 main and 4 aux buses
- Separate 16-track tape monitor
- Solo in place
- Outboard power supply
- Easy remix without repatching


M-50 CONSOLE

Series 50
M-50 Console

- 12 x 8 x 8
- Solo and PFL
- Built-in RIAA phono pre-amps
- Built-in direct boxes for instruments
- Balanced and unbalanced inputs and outputs
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Otari mixers and tape machines, for example).

Sometimes it is possible to intermix pro and semi-pro without an interface adaptor. A pro, transformer-balanced output can feed an unbalanced input without any signal loss, simply by tying one side of the balanced line to ground. If the pro equipment is equipped with an active balanced output, the load must be connected between one side of the balanced output and ground, leaving the other balanced terminal free. If this configuration is not adopted, the connection will short one of the output amplifiers to ground. It must be remembered that proper operation of the active output to unbalanced input will result in a 6 dB level loss.

AC Power Considerations

As a general rule, equipment in the same room can use unbalanced circuits so long as everything is fed from the same AC supply. Unless you are certain that other rooms are also on the same AC feed, any signals connected between rooms should use balanced circuits.

Most buildings are wired for 120/240 volt service, where a center tapped 240 VAC feed is brought in, and the 120-volt circuits are evenly divided between each half of the feed and the power line neutral. If part of the equipment is connected to one side of this feed, and the rest to the other half, hum-inducing current will flow through the interconnecting audio shields. When using unbalanced circuits, there is no way that this ground-loop path can be broken, which is why all equipment in an unbalanced circuit *must* be on the *same* AC feed. Using balanced circuits, the ground loop can be broken by grounding one end of the shield, and letting the other end float free. Any difference in potential between the equipment will appear as common mode voltage on the balanced audio line, and will be cancelled by the input stage.

With careful system planning, the use of expensive transformers can be reduced to a minimum, unless the RF suppression capability of balanced lines is required. For example, an unbalanced program output can stay unbalanced until it reaches the input transformer on a console in another room. Likewise, a transformer-balanced feed entering an unbalanced room may work without an additional transformer.

High-Impedance to Low-Impedance Conversion

A high-impedance, unbalanced output, which is usually found on most

PRACTICAL DESIGNS FOR THE SEMI-PRO/PRO INTERFACE

by Hank Landsberg, Drake-Chenault Enterprises, Inc.

Sooner or later, every broadcast facility engineer is faced with the problem of correctly interfacing a "consumer/hi-fi" cassette recorder, reel-to-reel deck, or what have you, with a typical studio installation. While most of us cringe at the thought of any consumer equipment being incorporated into a professional recording system, there are times when it's the only way to get the job done. Have you ever tried to buy a cassette recorder that has balanced outputs, capable of driving a 600-ohm load to +24 dBm? They don't exist; if you need to make cassette dubs, you are stuck with a consumer cassette unit, like it or not.

There are basic requirements that must be met in order to interface IHF standard, or "consumer" gear with professional equipment properly. Generally speaking, they are: Impedance matching, level compatibility, and input/output line topology.

To correctly match a studio "line output" with an IHF "line input," three criteria must be met:

a. The balanced studio line must be interfaced with the unbalanced IHF line input, without grounding either side of the studio line.

b. The balanced studio line must *not* be terminated by the IHF device or interface device.

c. The audio level of the studio line must be lowered by about 14 dB in ensure compatibility with the IHF device.

The circuit shown in Figure A accomplishes these goals, without transformers, in a very cost effective manner. The input impedance of the buffer amplifier shown is 40 kohms, so the studio line is not loaded. Because the amplifier is a differential input, neither side of the studio line need be grounded for proper operation. The choice of feedback resistor around the IC creates the needed 14 dB gain drop, so that the studio line level of +4 dBm is lowered to -10 dBV (300 millivolt), the IHF standard. Because a bi-polar supply is used to power the amplifier, the output of the IC is centered at 0 volts, hence no output coupling capacitor is needed.

To correctly match the output of an IHF standard device to a studio line input, three similar criteria must be met:

a. The unbalanced IHF output must be converted to a balanced line output.

b. The high impedance IHF output must be converted to a low output impedance, capable of driving a 600-ohm studio load.

c. The -10 dBV IHF output level must be amplified to +4 dBm studio level.

The circuit shown in Figure B meets these requirements efficiently. The input impedance of the line amplifier is 10 kohms, to prevent loading the IHF source. The feedback resistor around to first amplifier stage provides the 8 dB gain necessary to raise the audio level; the second stage operates as a "mirror" to provide the second half of the balanced output. This second output stage also creates another 6 dB of output level, so that the 8 dB gain of the first stage added to the 6 dB "free gain" of the mirror stage will produce 14 dB gain; a -10 dBV input will therefore yield +4 dBm output.

The output amplifier is also powered from a bi-polar supply; no output capacitors are needed. The amplifier clips at +26 dBm, allowing 22 dB of headroom over a nominal output level of +4 dBm. The output stage will drive a 600-ohm load to this level, and it is not affected by capacitive loads.

A typical IHF recorder interfaced with these two circuits will perform as if it were legitimate studio equipment! For a quick and inexpensive regulated power supply, *APfB* readers might contact Ault Incorporated — (612) 560-9300 — who make a "plug in the wall" bi-polar 15 volt supply that's capable of 150 milliamp load current. For portable operation, these circuits can be operated from batteries; a pair of 9-volt transistor-radio batteries will suffice if a maximum output level of about +18 dBm is sufficient.

Figure A

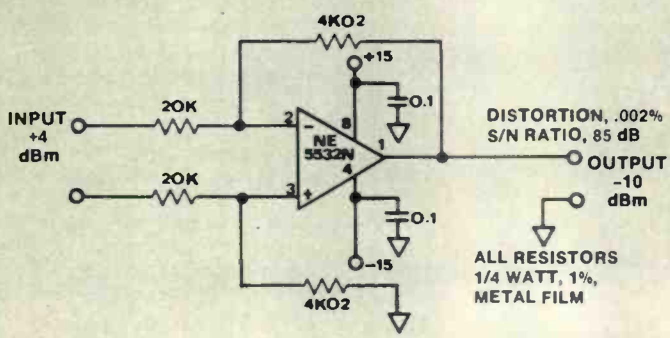
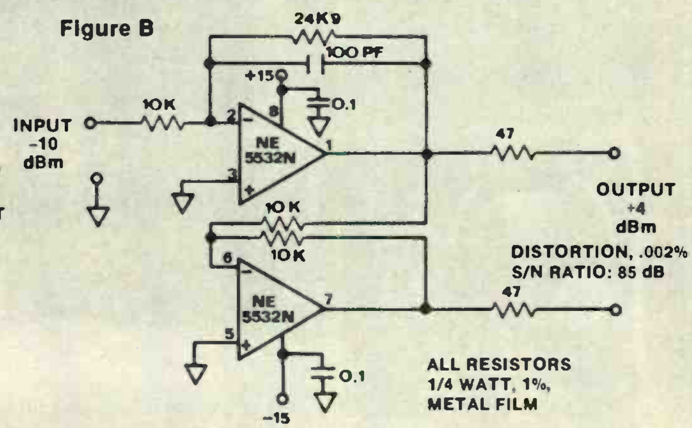
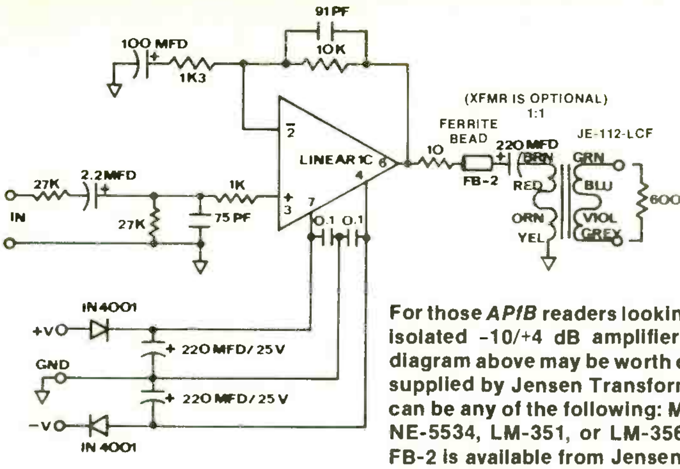


Figure B





For those *AP/B* readers looking for a transformer isolated -10/+4 dB amplifier stage, the circuit diagram above may be worth considering, kindly supplied by Jensen Transformers. The linear IC can be any of the following: MA-332 (preferred), NE-5534, LM-351, or LM-356. The ferrite bead FB-2 is available from Jensen Transformers.

semi-pro equipment with phono connectors, does not have sufficient output current to drive a balanced input. As a result, any attempt to directly connect the high-impedance output of a semi-pro tape deck, for example, to the low-impedance, balanced input of a mixing console can cause frequency response and distortion problems. The only exception could be the case of a balanced bridging input, if it offers enough gain to compensate for the lower level output from the semi-pro piece of equipment.

Circuits to provide the proper gain and impedance matching between pro and semi-pro equipment are shown in Figures 1 and 2. The circuit in Figure 1 uses a resistive voltage divider to drive a differential amplifier, thus providing an active, balanced input. The output of this device can then be used to drive an unbalanced semi-pro input.

In active, balanced inputs the common mode rejection is dependent on the tolerance matching of components in each half of the circuit. Here the common mode rejection is set by strapping the positive and negative inputs together, and feeding a signal between them and ground. The CMR rejection control is adjusted for the best output null.

The circuit in Figure 1 uses op-amps in a bridging configuration to provide a low-impedance, active balanced output, and will match an unbalanced output to a balanced input.

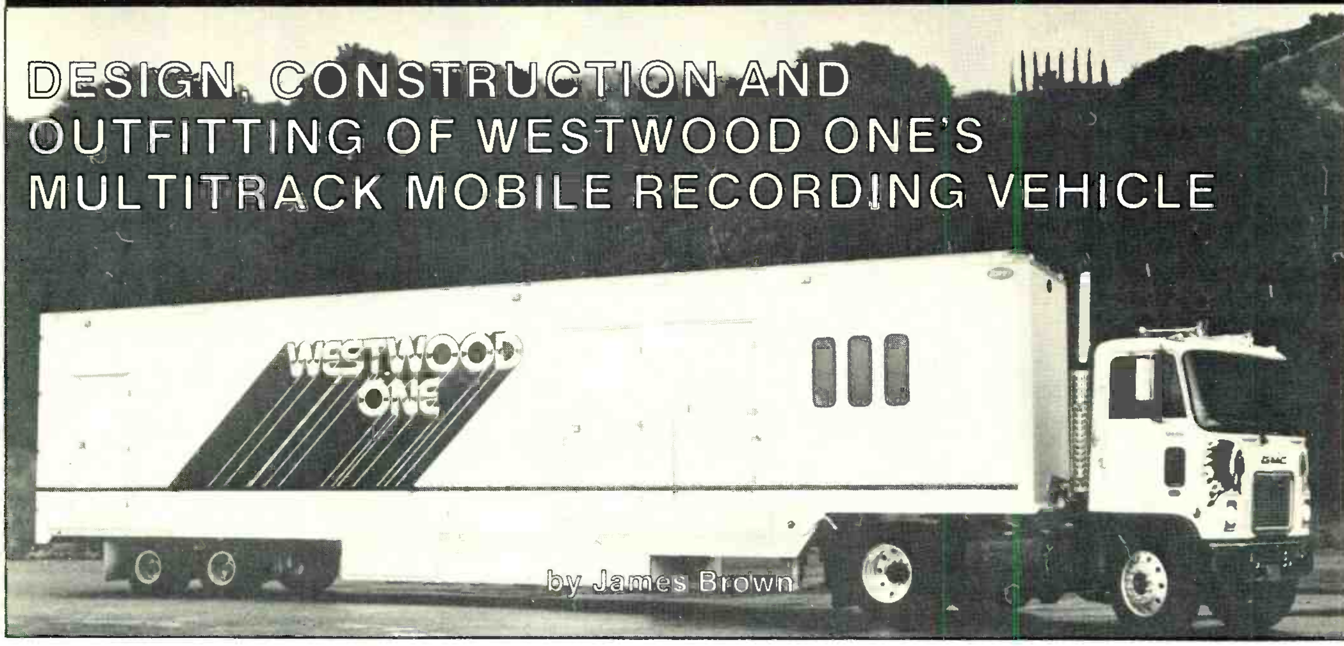
Transformers may be added to both Figure 1 and 2 circuits if desired, and generally provide for the best common mode rejection. Sometimes the common mode currents can get high enough to drive an active input stage into non-linearity.

Several commercial interface devices are available for proper connection of pro and semi-pro systems. Such units can be permanently installed for patch-bay access whenever a piece of equipment from the outside is brought into the studio.

In all cases you must first determine the nature of your device. This can be found in equipment manuals or, in many cases, by simply looking at the connector. Phono connectors almost always mean unbalanced circuits, whereas an XLR input or output may be deceiving, although the presence of XLR connectors usually signifies operation at around 0 dBm. Whenever uncertain, check with the manufacturer or dealer.

With proper care and feeding, a production studio should experience no trouble mating balanced and unbalanced systems, and should be able to operate in an environment free of annoying hums and buzzes. ■■■

DESIGN, CONSTRUCTION AND OUTFITTING OF WESTWOOD ONE'S MULTITRACK MOBILE RECORDING VEHICLE



by James Brown

When Westwood One was founded seven years ago, national radio was in a state of suspended animation. The major networks had become little more than national news services, cranking out headlines on the hour and half-hour. And while a handful of syndicators supplied programs to radio stations across the country, it was a haphazard process of demand, and not much supply. In other words — those were the "Bad Old Days."

Today, however, national radio is booming, and companies like Westwood One deserve much credit in its evolution from sleepy village to thriving metropolis. Now Westwood One is arguably America's largest producer and distributor of nationally sponsored radio programs — with 28 national programs that air on more than 3,000 stations. Its productions range from such 2½-minute features as *Spaces and Places*, *The Playboy Advisor*, *Star Trak*, and *Off The Record*, to series like *The Dr. Demento Show*, *Special Edition*, and *Rock Album Countdown*. And, lest anyone forget, live concerts.

Live Concert Production

During 1981, Westwood One produced more than 100 concerts for national distribution; in 1982 that number increased to 150 concerts. Its country concert series, *Live From Gilley's,* is heard on more than 400 stations; *In Concert* airs on 250 rock stations; *Coca Cola Super Star Concert* is taken by 350 stations; while *The Budweiser Concert Hour* is broadcast on more than 100 black, urban and R&B stations.

As such, Westwood One has become possibly the largest single remote broadcast client in the world, utilizing just about every major remote recording vehicle in the US and Canada. Over the years, Westwood One has done business with such mobile recording vehicles as the Wally Heider Truck, Record Plant Mobile in New York, Le Mobile in Montreal, Fanta Recording out of Nashville, Recording Connection in Cleveland, and Artisan Recording based in Miami.

"In 1981 we spent more than $1 million renting mobile recording vehicles," Westwood One founder and president Norman Pattiz says. "At that point we felt it was time to figure out a more cost-effective way of doing business, and to ensure that we could control the destiny of our product from start to finish."

Thus, Pattiz made the decision to build Westwood One's own custom-designed mobile recording studio — a 45-foot, Hi-Tech unit with state-of-the-art audio equipment, acoustically comparable to any recording studio in the world, with a lounge large enough to accommodate eight people.

The decision to build came after many months of looking at existing units to see if they would fit Westwood One's needs. They didn't.

"We looked around and everything out there was basically spartan in nature," says Richard Kimball, Westwood One's director of concert programming. "Because we do business with a lot of recording artists and managers, we wanted a mobile unit that was not only technically sound, but one that could incorporate some of the creature comforts we were looking for."

"There also wasn't any grand design for the construction of these units," adds Brian Heimerl, Westwood One's operations director. "They were all customized to fit the needs of the people who used them. They were built by and for their *operators*. So that's when we gave up trying to find something out there that was right for us, and decided to go out and build one ourselves."

Custom-Designed Mobile Studio

With a budget of $500,000, Westwood One then went about the business of assembling a team to build the mobile unit. Recording engineer Biff Dawes, an alumnus of the famed Wally Heider Recording Studios, Hollywood, was joined by maintenance engineer Dave Faragher, remote manager Doug Field, and audio engineering consultant Dave Brand to take on the task. They were given six months to do the job.

"These are people who've spent a lot of years doing remote broadcasts," Pattiz offers. "They knew what we wanted to do, and had the experience and expertise to build the studio, from the ground up, in six months time."

One of the very first orders of business was to conceptualize the design of the unit itself — a 45-foot trailer that would house the studio equipment. After much shopping around, the team settled on Coachcraft Engineering, designers and builders of custom mobile homes, coaches, studio dressing rooms, and makeup vehicles. Although Coachcraft had never built a technical vehicle such as this one, the company's workmanship was considered superior to the others that had.

"Again, there was no real industry standard for building these units," Pattiz says. "So, since Coachcraft did

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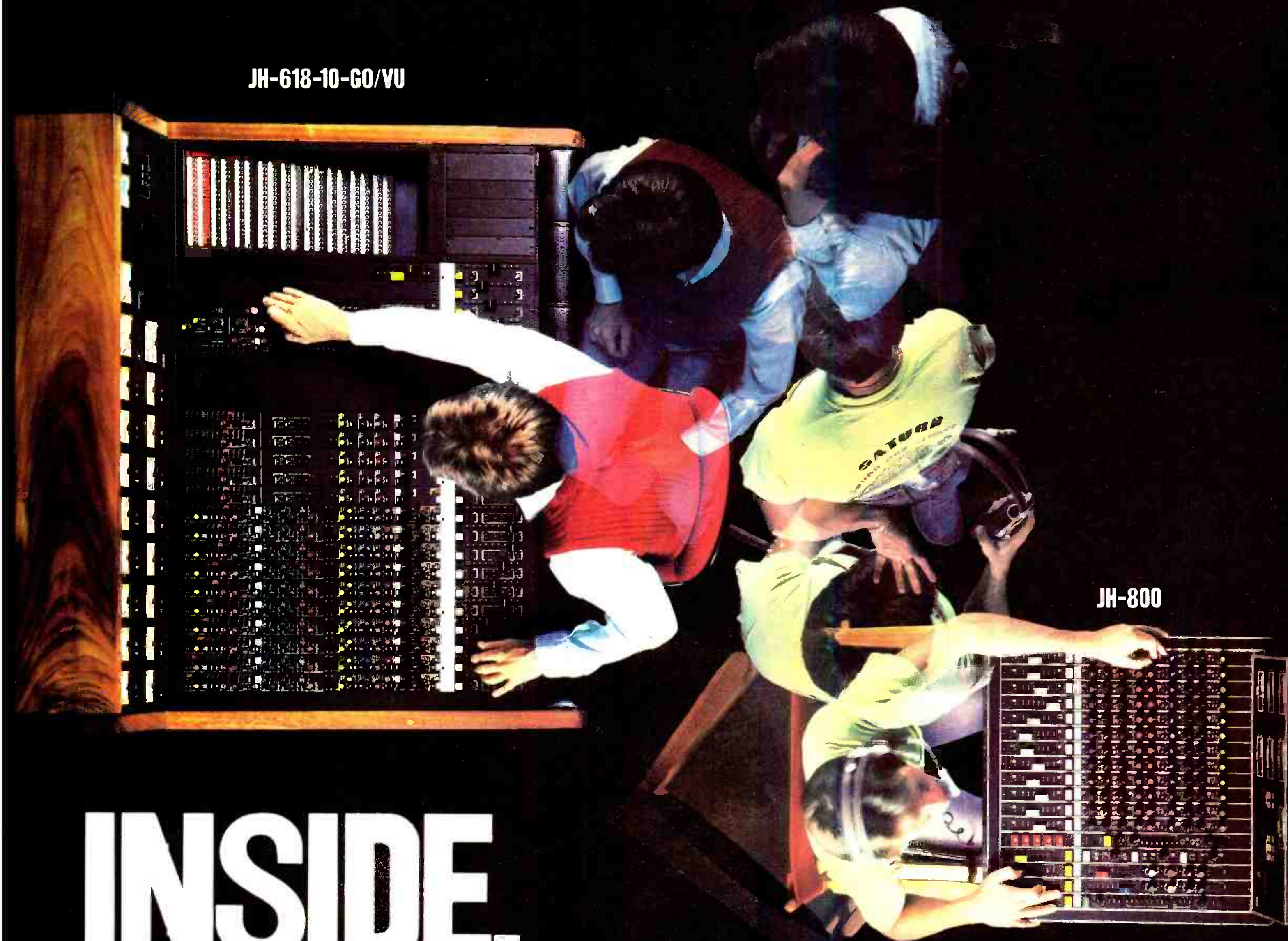

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MCI JH-600 console fitted with 36 inputs, routing to 24 busses.

exceptional work, we decided to create the standard ourselves."

With construction of the trailer underway, next came the work of assembling the equipment to occupy it. As with any custom-designed entity, the equipment and features were chosen specifically for the job at hand. Biff Dawes, a 10-year veteran of recording live albums with the likes of Fleetwood Mac, the Eagles, Devo, Tommy Tutone and many others, was a major force in determining which pieces of equipment were chosen, and how it would all be assembled. Also actively involved in the supervision of construction and design of the mobile studio was Jim Seiter, formerly director of remotes for Wally Heider Recording, and now affiliated with Sye Mitchell Sound Company.

Recording and Production Equipment

Westwood One's mobile recording studio is equipped with an MCI JH-600 Series console with 36 input channels routing to 24-track outputs. There are two Ampex MM-1200 24-track tape machines, an Ampex ATR-102 two-track mastering machine, a Sony color video monitor system that includes remote control camera with zoom lens, and a main speaker monitoring system that consists of two Altec 604-E drivers in custom DeMedio cabinets fitted with Mastering Lab crossover units.

Additionally, the studio is equipped with a full complement of outboard equipment set in a series of convenient racks, and capable of providing support for every possible live recording and mixdown situation. Full limiting and equalizing capabilities are provided, together with an Eventide Harmonizer effects and delay unit, an AKG BX-10 echo unit, full intercom system, and a Sphere Model 1604 sub-mixing system. The control room can be isolated from the lounge by closing the pneumatic soundproof door.

Dawes explains the equipment selections: "The MCI console is built specifically for remotes," he said. "With 36 inputs, it's small, the modules are smaller, and it's made to fit in the truck. It's also built for durability and stability. We've had *real* good success with it."

The Ampex MM-1200 multitracks, Dawes says, "were chosen for size and reliability. I'd worked with them at Heiders [recording studio] for some time and liked the sound. It's a 'punchier' sound."

Dawes considers that the mobile's custom Altec speakers, combined with a unique, acoustically corrected room design, make it possible to mix in exactly the same environment — and on the same equipment — that the event was recorded.

"The truck was too small for a time-aligned system," he continues. "This system is one that duplicates the old [Wally Heider] Mobile Unit One sound that the Heider truck was famous for."

— *continued overleaf . . .*

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The interior of the control room was constructed to not only offer a continuity of design, but to ensure the best possible acoustics. "The walls in the control room are contoured to keep away from 90-degree angles, as is the ceiling," explains Dave Faragher. "It changes angle ever-so-slightly to break up the flat ceiling reflections. The ceiling also changes materials in alternating panels.

"The interior finishes in the control room are carpet, fabric, rubber tile, formica and technical surfaces. All the technical surfaces have been placed so as not to interfere with our acoustics."

Another advantage to the control room, says Dawes, is that "you can move the console away from speakers for better monitoring. In some of the other trucks, where the monitor system is right up against the console, you couldn't hear accurately. Here, some people have come into the control room and commented on the 'wasted space.' Well, for monitoring, the most important thing *is* to have that extra space."

One of the other features considered unique to the new Westwood One unit is a pneumatic levelling system that can allow one person to level the truck under even the worst surface conditions. Also, a microphone input panel system is located inside the control room, away from any weather problems and potential vandals, and offers easy access to mike lines during the actual production.

A custom-built, 32-pair stage splitter system — originally laid out and designed by Best Audio — allows Westwood One production staff to split mike feeds with total isolation for the truck and PA system. It is, says Faragher, "a compilation of all other splitter systems' best features."

The microphone splitter boxes — intended to provide parallel and isolated outputs for a band's sound reinforcement system, and the mobile vehicle's mixing console — feature Deane Jensen isolating transformers, AMP connectors which, according to Faragher, are compact, mate more positively, and provide long durability in such applications, and a rugged case construction for "greater confidence in those situations where you never get the chance of a second take of a live performance."

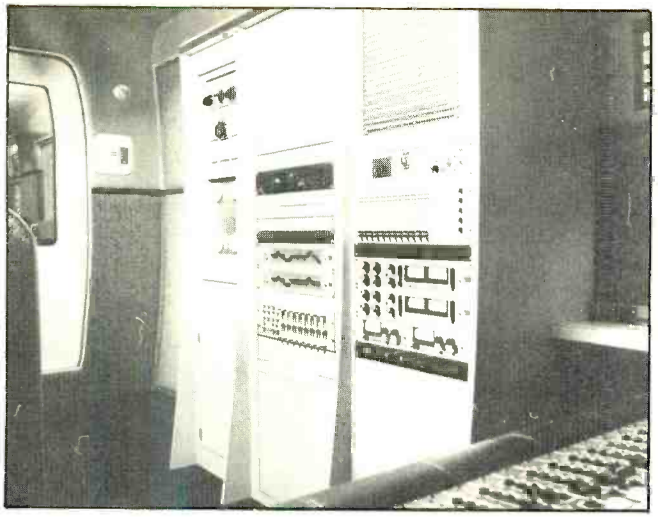
There was also a very good reason why Westwood One decided to enclose all of this audio technology in a trailer rather than a tractor-trailor rig, or panel truck chassis. "We made that decision based on the fact that when other vehicles had a problem, breaking down for one reason or another, that was it," Pattiz explains. "It was out of service. So, by building a trailer, we could choose whatever drive train we wanted, and also have a much larger area to work in. At 45-feet, it's the longest legal-sized trailer on the road."

Separate Lounge

The custom-designed lounge at one end of the trailer can, as mentioned, can accommodate eight people, and provides separately controlled audio and closed circuit TV monitoring. Also, the area can be completely closed off from the control room with a pneumatic door, thereby becoming a separate room of its own. The lounge's fixtures include audio monitors with separate level controls, a separate TV monitor to keep an eye on the stage or to view commercial broadcasts, and electronic bar, and refrigerator with icemaker. Its own entrance to the outside world ensures that the control-room activities remain undisturbed during a remote recording date.

While the lounge is built for comfort, it does have its functional side as well. As Richard Kimball points out, "For instance, if for one reason or other we lost a guitar track on stage because of a faulty pickup or microphone, we could put the guitarist in the lounge, and overdub on the spot. Or, if we were doing a satellite broadcast and wanted an off-

Central area: pair of Ampex MM-1200 24-tracks



Equipment rack: EQ, compression, and reverb units

stage announcer, he could use that room as well. There's an input panel for both voice-overs and overdubs, and the lounge is as acoustically sound as the control room."

Since its delivery in late June of last year, Westwood One's mobile recording studio has recorded some of the most popular groups in the country — Foreigner, Journey, Quincy Jones, Sister Sledge and many others — as well as providing audio support for many video productions. When not in use by Westwood One, the truck is also available for rental.

Norm Pattiz estimates that his half-million-dollar investment will "pay for itself in two years," but says that if the demand for the unit's services increase, it could be less than that.

"The studio has literally not been idle since it was delivered to us," he confides.

But perhaps the finest compliment of all came when Stevie Wonder — a man who knows more than a little about good music — was given a tour of the new studio. Wonder was said to be not only impressed by what he heard, he ordered a studio for himself just like it.

"When someone like Stevie Wonder pays you that kind of compliment," says Pattiz, "then you know you've done it right."

GBH MOBILE UNIT 4

CUSTOM-DESIGN FOR HIGH QUALITY AUDIO RECORDING AND POST-PRODUCTION

by Paul D. Lehrman

The GBH Mobile Unit 4 contains tandem API 40-input by 24-group split console, two Otari MTR-90 24-tracks, two Ampex ATR-102 two-track tape machines, and a respectable amount of monitoring and processing gear. Recording equipment is laid out inside a 40-foot former Greyhound bus, which was built 12 years and a million miles ago by MCI, a Greyhound subsidiary (and no relation to any of the other various MCIs). The truck not only handles WGBH's own programs, including *Evening at Pops* and *Evening at Symphony*, but also has done work for television companies —USA Cable, Metromedia, and Star TV; DIR and NPR radio networks; and even conventional record projects. The unit is used for both recording and post-production: "It's the best mixing facility we have," says operations supervisor Steve Colby.

GBH Production Services, which owns two video trucks as well as the audio bus, is actually a separate company from the WGBH Educational Foundation, Boston, which runs three television and an FM radio station. The operation was set up to make money, partly in response to government cutbacks in funding public broadcasting. The video trucks spend a lot of time with such clients as ESPN, the cable sports network, while about 60% of the audio unit's time is devoted to outside commercial projects.

Flexibility a Key Factor

Colby, the moving force behind the unit's design, explains that it was the need for flexibility in handling such a wide variety of work that determined many of the unit's design decisions. "When we started to work on [the truck]," he says, "we 'invented' a production scenario for an FM-stereo/TV simulcast, complete with multiple talent locations and several split feeds, as the most complicated job we would face. Then we came up with an equipment list — hardware the station had already, and others that we would have to buy — and arranged it on paper so that the console and tape operators would have the best access to the equipment, without falling all over each other. All of this would have to fit into a space no more than eight feet wide, because of the regulations concerning what can be driven on US highways. We determined the optimum dimensions of the space, and then went one step further and determined the minimum dimensions.

"We looked at trucks, trailers, mobile homes, and buses, in terms of cost, storage space, suspension, and a lot of other factors. We figured out that a bus would have the maximum amount of easily-converted space inside, and the luggage bays underneath would let us store external equipment in such a way that we could get at it without disturbing the operators inside. While the stage crew is unloading cable, we can be inside setting up the console and tape decks.

"Buses are also easy to service — Greyhound and Trailways have seen to that — and, if we break down, any '76' station has got the right parts, and a good mechanic."

The unit's 24- by 7-foot control room contains a pair of ganged API consoles. A 24-track board already was owned by the station, while the add-on 16-input desk came from a studio in Toronto. The two consoles are set up at right angles to one another, with the larger board facing the UREI 811A Time Align Loudspeakers, and Sony video monitors at the back of the bus.

The two API consoles usually are connected together, for 40-by-24-by-stereo operation, but can be operated independently for split-stage venues. The smaller desk is used mainly for "set 'em and leave 'em" signals, such as drum miking. Built into the larger board, for easy access, are eight dbx Model 903 compressor-limiters and one Model 162 stereo limiter, two Lexicon PCM-41 digital delays, and a Lexicon 224 digital reverberation unit.

To the operator's left, across the center aisle from the smaller board, is a rack containing a Nikko FM tuner and cassette deck, and 24 channels of Dolby noise reduction. On straight music recording jobs, the pair of Otari multitracks run at 30 IPS with no noise reduction, but for television the machines run at 15 IPS through the Dolbys, so that the crew doesn't have to worry as much about tapes running out during longer sessions.

Pair of ganged API consoles, Otari MTR-90 multitracks, UREI monitors, and outboard signal processors.

Next to the Dolby rack, towards the front of the bus, are audio distribution amplifiers, intercom and telephone systems, and a timecode distribution system. Here, too, is an extensive patch bay, through which all incoming and outgoing audio feeds can be accessed. Further towards the front is the tape operator's postion, with the two Otari MTR-90 24-tracks on one side of the aisle, and two Ampex ATR-102 stereo decks, plus space for another machine, on the opposite side. Storage bays for tape, maintenance equipment, and office supplies are located above the multitracks.

Isolated Area

Just behind the bus driver's position, separated from the control room by a soundproof birch-panelled wall, is an 11- by 7-foot lounge area, sufficiently large to seat six people and isolated enough to be used as an announce booth. "We were pleased we could fit it in," says Colby. "Besides allowing the crew to eat dinner somewhere besides backstage, or in some alley, it's a good place for producers and production assistants to hang out." A built-in desk can function as an interview table, a production-meeting desk, or a dining area. The lounge also contains audio and video monitors, a telephone, an ice chest, more storage space, and vents for the unit's upgraded heating and air-conditioning systems.

What had been the luggage compartments of the bus now are used for external equipment storage. Easily accessible from the outside of the vehicle are a custom-built 40-channel, three-way mike splitter, with Jensen transformers, and a powered cable reel that holds a 250-foot snake containing 40 shielded audio pairs, two intercom circuits, two coax lines, and four utility lines, all terminating in AMP multipin connectors. The line is filled with mica dust to keep it flexible. Although the snake is not usually called upon to handle video it can, with complete isolation from the audio signals. There are also various extension cables, mike boxes, AMP-to-XLR adaptors, a 15 KVA shielded power isolation transformer, and a 200-foot AC cable, also on a powered reel.

As a rule, the bus carries no SMPTE timecode equipment, but instead usually takes code, as well as vertical drive, video intercom, and two-way audio, through a 150-foot "umbilical cord" from whatever video unit it is working with.

"Video trucks always have SMPTE equipment permanently mounted," Colby explains, "so it's more convenient for them to generate it." Of course, the station has a wide variety of SMPTE readers, generators, and synchronizers, any of which can be brought aboard on short notice. Part of the truck's patchbay is a timecode distribution system, which is well isolated from the audio wiring, and can feed incoming code to the multitracks, the two-tracks, or any other audio or video equipment that has been brought along.

For television post-production work, a four-track Ampex is brought in, along with a pair of Dolby 361 noise-reduction racks, and a SMPTE reader/generator. The finished tape, which consists of two Dolbyed audio tracks, a guard band, and a timecode track, then can be laid back on to the master video tape in WGBH's editing suite [described in the Spring 1982 issue of *APfB — Ed*].

Multicore snake, stage boxes, splitter/DI boxes, and cable reels.

Colby, with 10 years in broadcast audio under his belt, is an experienced music mixer, but how much input he has at any one assignment varies considerably. "When WGBH does classical programs," he explains, "it assigns a producer, and he and I work together to get the mix. On some rock shows, the client will leave it totally up to me as to how it goes down. We did one shoot for DIR Broadcasting and, because they were confident we could handle it, nobody from the company even had to come along with us. Other times, the client will assign a co-producer."

The bus travels with a large microphone inventory, which ranges from Neumann U87 and Crown PZM, to AKG D202 and Electro-Voice RE-15 models. "We could do all of our own miking," says Colby, "but usually we don't. At [Boston] Symphony Hall, we have a permanent system installed. We start with that and add to it as necessary. When we do a rock show, we'll take feeds from the house PA through our own splitter, which makes the artist and the PA company more comfortable. Sometimes we'll come across a drum mike in which the diaphragm has been beat to heck, in which case we'll replace it with one of our own of the same type."

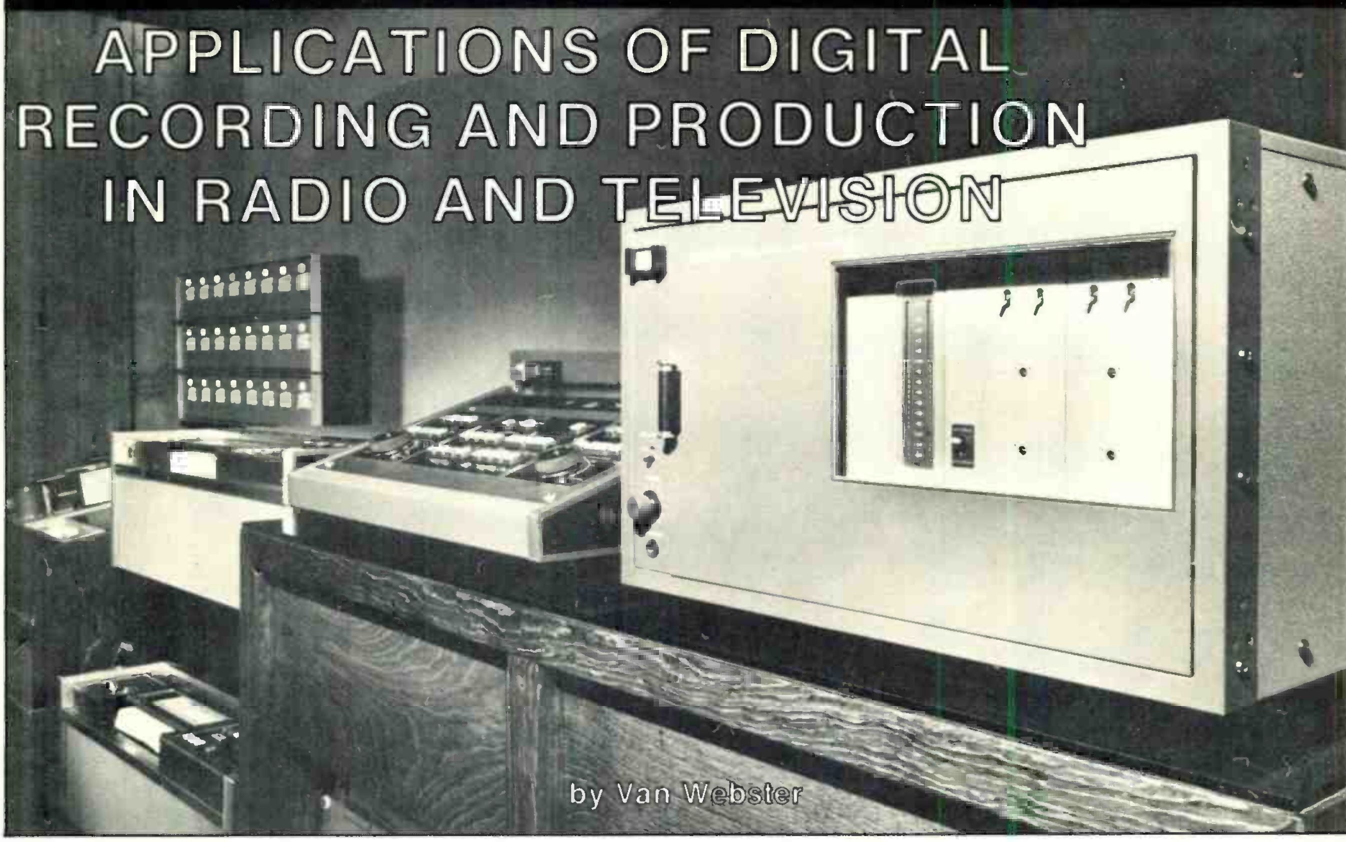
From drawings to production, the conversion and construction of GBH Mobile 4 took a scant six months. "TV stations live on deadlines," says Colby. "We had to be ready for an *Evening at Pops* shoot, and we were — by about 24 hours."

Speed remains an important consideration in the unit's way of operating. "We try to get in and out quickly, without bothering anybody," Colby offers. "I just wish that there was a standard layout for the multipin connectors; it would make our lives *much* easier. We have our own video truck setup so, when we're running with [that] it takes all of three minutes to hook the two units together."

Which may be one of the reasons why outside clients, even those who have their own mobile video facility, on several occasions have hired GBH's video unit as well when they book the audio bus. Steve Colby considers that the reasons go beyond that, however. "We happen to have some very talented video and audio crews, and some good facilities," he says. "The crews work well with each other, and they know their equipment. I don't know of another production house anywhere that has separate stand-alone audio and video mobiles. Having both of them belong to the same company helps a lot." ■■■

APPLICATIONS OF DIGITAL RECORDING AND PRODUCTION IN RADIO AND TELEVISION

by Van Webster



— The Future Role to be Played by High-Quality Digital Sound in the Broadcast Environment

Digital audio recording has been the subject of much technical development during the past several years. manufacturers have shown prototypes and have offered equipment for sale to the audio industry. More recently, hardware and software manufacturers have shown digital audio systems intended for the home consumer. All of this activity points to a concerted effort by equipment manufacturers to make digital equipment a viable part of the audio production marketplace.

Digital audio recording is a process which converts audio range electrical signal into binary code, which can be stored, duplicated, and recovered easily. As has been stated before, the principal advantage of digital recording is that the distortions associated with conventional analog recording are eliminated in the storage process. In addition, any transmission medium which has sufficient bandwidth (i.e. broadcast television), may be used to transmit digital data without the usual limitations and losses associated with conventional audio distribution.

Presently, digital equipment is most often being used to produce hybrid digital/analog products such as records, and for the preparation of soundtracks for feature films. In addition, some pioneers have used digital audio in the production of soundtracks for television commercials. However, the use of digital audio recording and transmissions is in its infancy, and many applications have only been touched upon. For the broadcaster, the advent of digital audio promises improved audio quality, and greater listener acceptance of prerecorded programs.

Digital Audio in the Production Environment

Given the present characterisitics of broadcast audio, including limited bandwidth and dynamic range, and the poor quality of receivers (especially television speakers), the advantages of digital audio for broadcast might at first seem questionable. Why, for instance, would a broadcaster need a 90+ dB dynamic range for AM radio?

The answer lies in the advantages of digital recording for audio production.

— the author —

An "audio-visual futurist and computer buff," **George V. "Van" Webster** is considered by many to be a pioneer in digital audio recording, and in producing computerized audio/visual presentations and commercials, through his twin companies Digital Sound Recording, and Hope Street Studio, both based in Los Angeles.

Most broadcast production requires multiple generations of program material before it reaches the air. Digital recordings may be duplicated virtually limitlessly without generation loss. Each duplication regenerates the binary code, correcting for errors, without the expected build-up of noise and distortion that is associated with analog recording.

Another significant application is the use of digital audio processors to encode and decode program material transmitted from a remote location, such as a live concert broadcast. Often the limiting factor in the quality of remote broadcasts is the transmission link between the remote location and the broadcast transmitter. By using data lines, a microwave link, or even a satellite relay, digital audio can effectively eliminate virtually all of the audio limitations of conventional analog relays.

Most of these techniques are just now being explored, and virtually all broadcast work in digital audio is pioneering in nature. There is nothing inherently more difficult or complex about the

application of digital audio production techniques but, because it is unfamiliar to most technicians, each application will require some extra care and time to work out the production details. In addition, the use of digital audio equipment may point up equipment problems that have been masked by conventional analog technique. The curing of these problems can improve the overall quality of the broadcat signal.

Digital audio has tremendous promotional advantages to the professional broadcaster. Because of its pioneering nature, and superior audio quality, broadcasters using digital recording can promote their concern for the listener and their program quality by publicizing their digital audio work. There has been considerable publicity in the consumer press about developments in digital recording. Skillful promotion can tap in on this already established audience awareness.

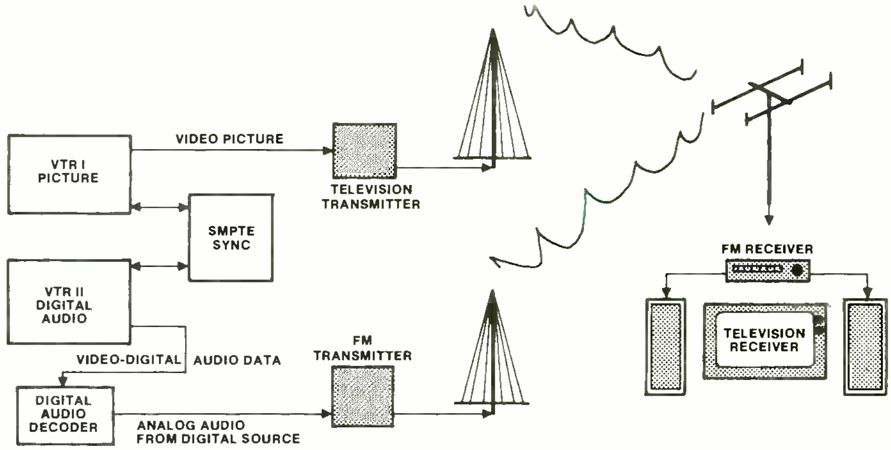
At the present time, the purchase of digital audio equipment may be prohibitively expensive for many broadcasters. However, digital equipment and technical expertise is available for rent in most major markets. The cost of digital recording and production must be weighed against the promotional and audio values received. For broadcasters who pride themselves on offering their audiences the finest quality, digital audio can be profitable. From a production standpoint, it takes aproximately the same time to create a program in digital audio as it does a conventional recording.

One possible exception may be editing of digital material. Many of the currently available digital audio systems use electronic editing, which must be accomplished in real-time. For producers familiar with the video editing process, digital editing will be quite similar.

Production Techniques and Considerations

Audio production for radio broadcast includes the recording and assembly of a wide variety of program sources, including records, tapes, carts, live vocals, and special effects. All of these elements are re-recorded and combined to form the final product. The multigeneration approach to audio production is especially prevalent in radio commercials, and syndicated programs. The use of digital recording to produce this type of programming can substantially improve the audio quality, and increase the intelligibility of the message.

In working with digital audio as a production tool, the general audio approach is similar to conventional recording. Two principal advantages may influence the production technique. The first is the freedom to go multiple generations without quality loss. Such flexibility can be especially important for submixing, and the preparation of beds for multiple voice-overs. The second advantage is the dramatically



TELEVISION AND FM AUDIO SIMULCAST WITH DIGITAL AUDIO SOURCE

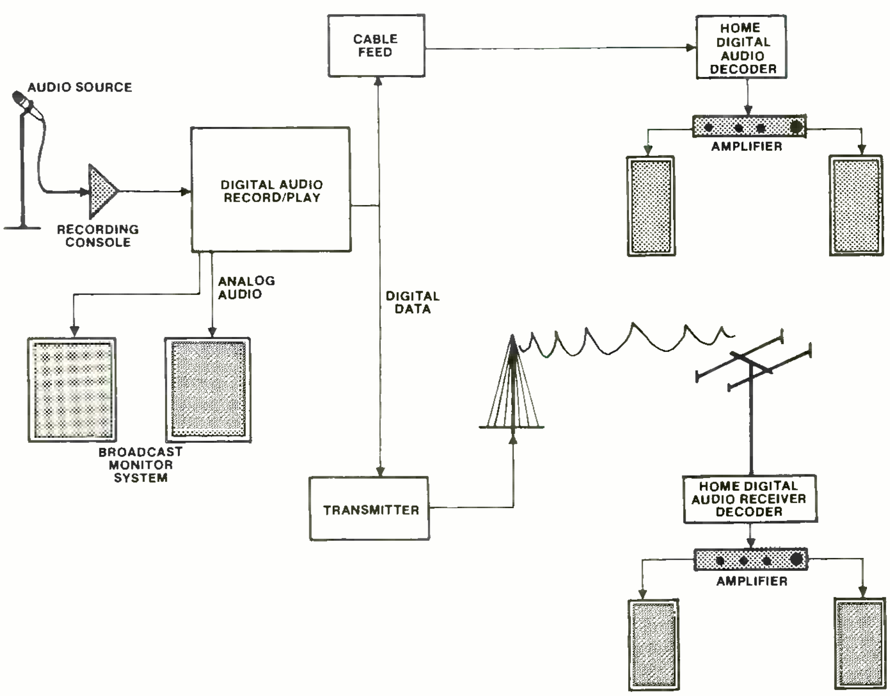
improved audio quality that can add presence and clarity to the program material.

Timing is a critical factor in all broadcast production. Because most digital audio systems will not permit off-speed operation (VSO), it is essential that the program be accurately timed as it is assembled. Speeding up the tape to reduce its playing time, or to raise the pitch of the music, is not normally possible once the program is in digital form. If such effects are desired, they must be accomplished during the transfer of the material from analog to digital.

Most digital audio systems have provisions for sophisticated electronic editing, which can permit a range of alternatives not normally associated with analog tape. Included in some systems are cross-fade capabilities, and highly accurate location of edit points. In addition, electronic editing allows preview of an edit before making the actual "splice." Electronic editing offers the further advantage that during the editing process the master tape is *duplicated*, and not physically cut. Because the master is never physically altered, it may be re-used and/or re-cut many times; such flexibility may encourage producers to experiment more with their program material. Given this freedom, however, the editing process may take more time than would be expected during conventional production.

Digital audio has been used sucessfully in the production of commercials. Most of the applications so far have been coupled with television productions. Original music and sound effects have all been digitally produced. Often commercial production includes the use of extensive audio processing to increase clarity and listener impact. The results of such extensive processing is a very complex audio signal that may be distorted substantially by analog

DIRECT DIGITAL AUDIO BROADCAST INTO THE HOME



DIGITAL AUDIO TECHNOLOGY

RECORDING AND PLAYBACK USING PCM ADAPTORS FOR BETA- AND VHS-FORMAT VIDEOCASSETTE RECORDERS

by Bert Goldman, KABL-FM Radio, San Francisco

Until now, analog recording has dominated the broadcast industry and the music business in general. While highly refined and of high quality in its own right, analog recording technology has certain, difficult to overcome, limitations, especially for the broadcaster. For instance, the frequency response of an analog tape recorder is subject to high-frequency loss due to head gap width and tape coating thickness, and low-range response below 20 Hz is affected by the effective head width. Also, analog tape dynamic range is limited by tape saturation — particularly in the high-frequency range — and by tape hiss.

Some attempts to overcome these weaknesses include raising tape speed, using high-output/low-noise tape formulations, and employing noise reduction systems. However, limitations of the analog system are not easily overcome. Performance of any analog tape system is determined by the cumulative effect of mechanical precision, tape quality, recording head, motors, setup, and so forth. Even with the high quality of today's best tape decks there is not much room for improvement in the basic areas, and we are still left with the fundamental drawbacks of analog recording. Recording, and hence playback quality, is directly affected by irregularities in the tape coating, dropouts, wow and flutter, and print-through after storage.

Digital recording, using pulse code modulation (PCM) technology departs significantly from the analog method. Digital electronics sample the signal at one of two "standard" frequencies: 44.1 and 48 kHz. Each sample is quantized directly into a discreet number, and converted into a binary code for placement on to the tape medium. Which, in the case of most presently-available stereo PCM systems, comprises a conventional ½- or ¾-inch videocassette recorder. Since the information consists recording and duplication. Commercial producers have found that the use of digital techniques for mastering can produce cleaner air copies by duplicating from "clones" of the digital master.

Changing Audio Requirements

The radio broadcast industry is undergoing a change as markets alter, and competition increases. New technical developments include companding ("compression/expansion") systems for home broadcast use. The successful broadcaster of the future will have to offer his listeners something unique in order to be distinctive. Digital audio broadcasts offer one potential. As pointed out in the Spring 1982 issue of *Audio Production for Broadcast*, producers such as Minnesota Public Radio have made digital audio recordings for radio broadcst. In another example, the British Broadcasting Corporation found that listeners preferred listening to a digital audio recording of a concert, rather than a live broadcast, because the losses in the transmission link from the hall during the live performances were clearly audible by comparison. Some sponsors, particularly large energy and high-technology companies, may be more willing to underwrite the cost of a digital broadcast from a cultural event than a more conventional program, because of its special nature.

For producers of network and syndicated programming, the ability to duplicate digital audio without generation loss is a substantial advantae for program distribution. Satellite links can be rented now for low cost, and the use of digital audio to transmit programs worldwide has become an economic and technical reality. The monies saved in duplication can more than offset the costs of digital production.

Digital audio is also a valuable tool in video production, both for programs and commercials. While conventional monaural audio is typical in broadcast video, the presence of stereo home televisions and video recorders will create a consumer demand for improved audio quality. In addition, programs first introduced by broadcast eventually may be released as home video products. Music specials on television have used simulcast FM stereo audio successfully for years.

One factor that has become an increasing frustration to producers of stereophonic audio for television, is the lack of consistent phase orientation in video recorders. Often in a single edit bay, each of the dseemingly identical recorders will have different phase characteristics; this is especially a problem when the program being produced needs to be mono compatible. Out-of-phase signals will cancel themselves out, and seriously degrade the mono signal. Digital audio, by the very nature of its recording process, is immune from phase problems, and can be especially helpful on stereo/mono productions.

— continued overleaf . . .

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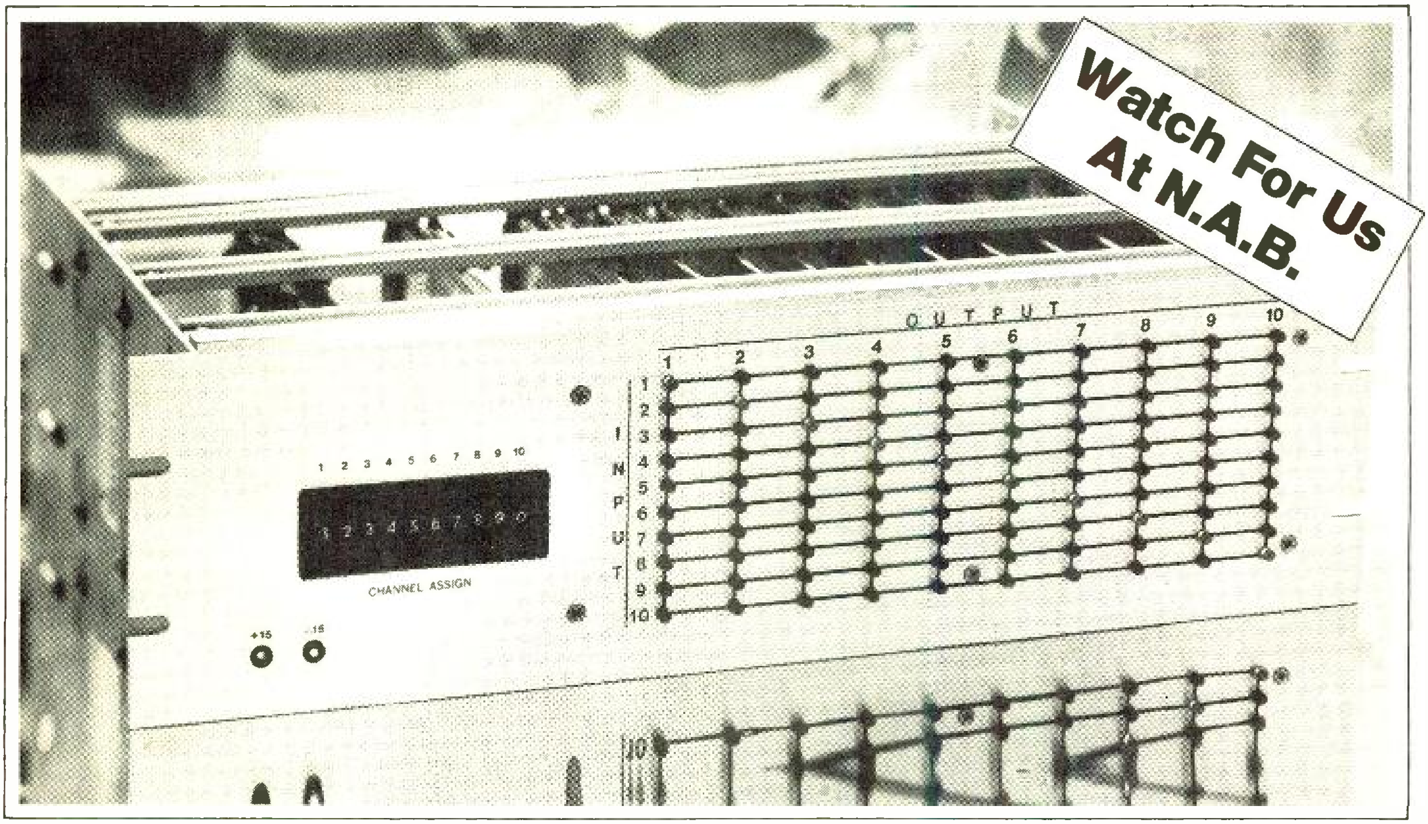

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Input Level	-10 to +24 db
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Output Level	+22 db Max (Adjustable 30 db range)
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Virtually all digital audio recording systems include a provision for SMPTE code synchronization. Some units use ¾-inch U-Matic broadcast videocassettes as the storage medium. Synchronizing such players is a simple matter for video editing equipment. The sonic advantages of digital audio can be especially apparent when elaborate video post-production involves multiple generations of video images. An analog audio track, which would suffer considerable degredation under such circumstances, can be stored in digital form on a separate video recorder. After the editing and effects are complete, the audio can be laid back to the final edited master with first-generation quality.

This same quality level can be maintained in video sweetening sessions where multitrack recorders are used to develop the master audio track. A mixdown to digital audio permits a synchronous, high-quality audio signal to be used in the final production. The use of an outboard noise-reduction system is unnecessary with digital recorders. Time-consuming audio alignment of two-channel mastering recorders is also eliminated, as is the use of outboard synchronizers linked to the editing controller.

Digital recording is ideal for a direct audio feed during a video simulcast. Using SMPTE synchronization, the digital audio recorder can output directly to an FM broadcast console. For best results, the digital audio signal should originate at the FM transmitter, since it will be degraded by land-line transmission. Sending SMPTE timecode over land lines as a synchronizing signal can enable separate locations for the audio and video sources.

Another alternative is to use a microwave transmission link to send video-encoded digital audio from the studio to the transmitter. A decoding processor at the receiving end can convert the signal to audio for broadcast. The microwave approach also can simplify the coordination of the video and audio sources.

Because digital recording is in its infancy, potential applications have just begun to be explored. The key to the successful development of digital recording is the availablity of digital processors in the home. When the consumer is able to receive and decode a digital broadcast signal, the quality of home entertainment will change dramatically. Conventional records and tapes will become largely obsolete, as interactive broadcasts allow people to make up their own programming in the home. Systems have already been proposed that permit home taping of digitally-encoded material with a license. If cable systems also are included in the broader category of "broadcast," then the possibility of digital transmissions becomes even more feasible. The key of its success lies in the marketing. Record and video companies will have to think of themselves as being in the home-entertainment business, and *not* just in the business of manufacturing and selling of objects.

On the production front, digital audio processors already are available which can be connected to ½-inch Beta and VHS videocassette recorders. Because such systems are frame synchronous, the digital recorder has the poter ial to replace the ubiquitous Nagra or high-quality portable analog tape machine as a field recorder. Electronic cinematography and the development of high-resolution television also promise to revolutionize the production of broadcast programs. Digital audio is a "high-resolution" system for bringing quality audio into the home, without transmission losses.

The audio production industry presently is in an era of rapid technological change. Digital audio represents one of the more dramatic of these new developments. In order for any technology to become commercially viable, both the marketing and the operations need to be developed. Broadcasters who want to be part of the future will recognize the experimental nature of digital audio techniques, and will include an "R&D" factor in the evaluation of the cost of digital audio. Digital recording may not be for everyone yet, but it seem certain that digital audio will play a big part in the future of broadcasting. ■■■

DIGITAL AUDIO TECHNOLOGY *continued*

of basically the presence of absence of a pulse, a digital system effectively can avoid the weaknesss of the conventional analog process.

Digital in the Production Studio

Used in a production sense, digital recording allows multiple generation recording with virtually no signal deterioration. Television simulcasts can be considerably simpler to synchronize, since both utilize the same video format — normally ¾-inch U-Matic VCRs on 1-inch videotape — and audio production houses can dub hundreds of tapes with the same quality as the master.

But, until now, digital recording has also had definite limitations, basically in the areas of cost and standardization. Over the last couple of years tremendous advances have been made in digital recording. The EIAJ 14-bit digital encoding standard seems to have been accepted by most manufacturers, thus addressing the standardization problem, and the combination of this standard recording technique with commercially available videocassette transports has drastically reduced the price of this new technology.

Ironically, by moulding this digital encoding technique with the commercially available videotape cassette systems, some exciting new advantages have become available, making this type of system even more attractive to the broadcaster. Possibly the most exciting is that of added controls. Since a pair of audio tracks have been recorded on the single video track, at least one, and sometimes two, analog audio tracks are now available to accommodate all of those great control functions that we've been mising for years in reel-to-reel machines. Controls utilizing primary, secondary, and tertiary tones like a cartridge machine may be added and used for precise cueing, and logging tones could be added

COMPARISON BETWEEN ANALOG AND DIGITAL RECORDING

	Analog (typical)	Digital (typical)
Frequency Response:	50 Hz to 15 kHz ±2 dB	2 Hz to 20 kHz ±0.5 dB
Wow & Flutter:	0.1% (with AC Motor)	Below measurable range
Speed Deviation:	±0.2%	Below measurable range
Separation:	—	85-90 dB
Dropout correction:	None	Parity detection/ interleaving
Signal-to-noise ratio:	68 dB at 7½ IPS	85 dB
Phase Distortion:	Dependent on setup, transport	Negligible
Power consumption:	150 watts	55 watts
Tape cost:	10½-inch metal approx. $20 each	T-120 VHS cassette approx. $15
Cue track:	None	1 or 2 analog tracks
Rewind time:	2 minutes, approx.	5 minutes, approx.
Record time:	90 minutes with 1-mil tape	120 minutes with T-120 VCR
Cost:	$2-$4,000 playback-only $3-$5,000 record/playback	$1,900-$3,700 record/playback
HeadLife:	3,000 hours	2,000 hours

Sony PCM-F1 audio processor, and companion Betamax VCR. Shown right is WFMT-FM Chicago's on-air studio, used to replay PCM-encoded concert recordings. Left to right: Sony's training manager Marc Finer, Sony technician Dennis Dougherty, WFMT engineer Larry Rock, producer Rich Warren, and announcer Steve Reeder.

for identification. Linked to a shuttle control with the proper interface system, these units may be random accessed.

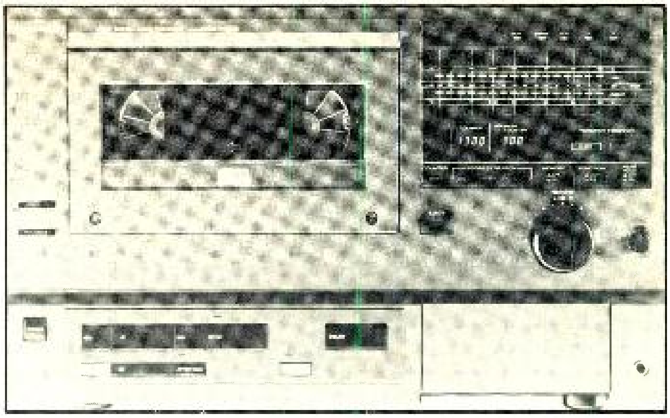
The cost of using PCM technology has truly crossed into the realm of possibility. The retail price of "consumer-type" PCM units has fallen well below $2,000, which is quite competitive with most quality reel-to-reels. The cost factor of the basic transport, however, is only part of the story. By utilizing commercially available videocassettes — Beta or VHS, as opposed to the more expensive U-Matic "professional" formats — the cost of the recording medium becomes competitive with analog. Because videocassettes are smaller and usually lighter than a 10½-inch metal open-reel counterpart, distribution and storage costs may be less. Also, since approximately two hours of program material can be recorded on to a videocassette, less transports are required, further reducing the ultimate cost.

Improved Audio Quality

Audio quality is perhaps where the use of the digital PCM system stands out most. With noise floors dropping rapidly with the advent of newer technology, the analog open-reel being used by broadcasters could be the weakest link in the chain. By the time the source material is mastered, dubbed, and played back, a 60 dB signal-to-noise ratio may be but a fantasy. Since PCM recording quantizes the audio into discreet binary values, noise is no longer a great concern, and figures of 85 dB signal-to-noise are typical.

The new PCM digital videocassette systems provide other audio quality advantages as well. Frequency response is razor flat to 20 kHz, with THD reduced to less than 0.01%. Dropouts, a nuisance at best in analog recording, could be devastating in digital record-

Technics Model SV-P100 PCM processor and VCR



Hitachi PCM-V300E, capable of mounting in 19-inch rack

ing. Therefore, a system called the Cyclic Redundancy Check Code was devised. The CRCC can detect 99.9985% of all dropout errors, and reconstruct the missing information. If CRCC fails, word interleaving and linear interpolation averages the values of adjacent correct data, and approximates the missing value. The result is virtually dropout-free audio. During PCM playback, any speed variations are corrected by locking on to a quartz reference before conversion to analog form. This method reduces wow and flutter to below the measurable range.

Another aspect of PCM recording is of prime interest to the broadcaster. Since the two-track audio information is not dependent on head azimuth or other time distorting situations, phase distortion is virtually eliminated.

So price, audio quality, and control features are big plusses in utilizing the new PCM Beta- or VHS-based videocassette systems in broadcasting. Now the bad news. The transports integral to the digital cassette record units, due to their helical scan process, tend to be considerably more mechanically complex than an open-reel transport. Servicing the mechanical linkages may cause many engineers considerable problems, both in time to service the units and in "psychiatric" care after the fact.

Editing on digital transports also can cause problems. Since the tape is enclosed in a plastic cassette shell, and the method of recording is considerably different than analog, different measures must be taken during the recording process. To overcome the problem of editing, special functions have been added to digital machines. Technics uses "Jump" and "Search" marks to skip or locate specific items, and Hitachi provides a digital mute function which records "zero" digit signals on the tape for editing.

Tape head wear, also due to the helical scan process, is limited to about 2,000 hours. Which means that with daily use of about 6 hours, we can expect no more than about 1 year's worth of service. There are offsets provided for this, however. At least one of the transport manufacturers provides a data loss readout on the front panel so you can tell if the video head is dirty or wearing down, and have plenty of advance warning to change it. Also, since there are basically no adjustments in the rotaing head assemblies, replacement of the head cylinder is a reasonably fast and easy procedure.

A final problem, and perhaps the hardest to overcome, is the videocassette system pre-roll time — the amount of time it takes from when the transport receives a start command to the point at which audio is enabled. This may take 5 or 6 seconds from a full stop position. The time could be reduced by leaving the transport in the "pause" mode, but the trade-off is increased head wear since the spinning head maintains contact with the tape in this position.

One way to overcome the problem of pre-roll in broadcst automation systems would be to bring out of the system the "This source next" line, which would place the machine in the Pause mode and ready to play.

Still, no system is perfect, and since it seems we're still quite a way from a "song-on-a-chip" technology, broadcasters may want to look at the PCM digital cassette system for the next generation of audio reproduction.

The following are three of the most popular PCM digital units currently available:

1) **Technics SV-P100.** Easily slips into a 19-inch rack, and uses VHS-type videocassette. Standard control features include a "Search" mark, which can be used to cue up a section. Also a "Jump"

mark may be recorded to skip over a part of the tape. According to Technics engineers, the audio track also may be accessed for additional uses. The unit also has a playback data check to assess head condition. The system is retailing for about $2,500, but prices have been reported as low as $1,900 in some locations.

2) **Hitachi PCM V300E**. This deck also slips nicely into a 19-inch rack. A VHS-based unit that offers a few additional features over the SV-P100, with a corresponding increase in price. It provides an address-search function and a "Data Slice Level Control," which enables improved tape interchangeability without having to make the occasional tracking adjustment. This system retails for just around $3,500, but Hitachi will substantially discount this price.

3) **Sony PCM-F1**.This unit requires that a separate videocassette machine be provided. Which may be a good idea since you can choose your own transport, but may also cause problems in packaging for the broadcaster. Sony offers 16- or 14-bit quantization. The former allows another 4 or 5 dB SNR and 0.005% THD, versus 0.007% with 14-bit. The PCM-F1 cost is around $1,850 without transport.

With the new-generation digital cassette recording units, a new era for broadcast reproduction may be near. The advantages are many, although there are still many hurdles yet to be overcome. But the new PCM cassette systems are now clearly competitive with conventional analog transports, and perhaps now is the time that we consider working this new technology into our broadcast chain. ■■■

WCLV EQUIPPED TO PLAY DIGITAL COMPACT DISKS

WCLV, Cleveland's Fine Arts Station, was one of the first radio stations in the country to equip itself to broadcast digital Compact Disks as part of its regular programming. A Sony CDP-101 digital audio disk player and a supply of the new digital Compact Disks were supplied to the station in late February.

In early March the station broadcast the opening measures of Richard Strauss' "Also Sprach Zarathustra" from a Technics digital Compact Disk. According to WCLV, it is the first station outside of New York and Chicago to be equipped with a Compact Disk player.

Since the Compact Disk is read by a laser rather than a needle as with the current analog method, this means the end of background noise, clicks, pops and hiss. Also, the records have a practically unlimited life, due to lack of wear. And the system takes full advantage of the superior sound quality inherent in digital recording.

WCLV plans to use the Sony digital system and CD records in regular programming, as well as on special programs devoted to highlighting this significant advancement in the art of sound reproduction.

CARE AND REPAIR OF REEL-TO-REEL TAPE TRANSPORTS

by Thomas L. Mann, Jules Cohen & Associates, P.C.

The use of reel-to-reel tape in broadcast production and programming is frequently in sharp contrast with its use in the recording industry. One reason for this is that broadcasters have varying end-uses for tape that necessitate differing quality levels, and often the need for compactness and convenience. Broadcasting is a "real-time" industry, and one in which immediacy frequently overrules quality consideration.

Some of the most frequent end-uses in broadcasting include: music recording —usually transcriptions from disk for more convenient airplay; commercials — frequently further dubbed to tape cartridge; news actualities from live or telephone interviews; and pre-recorded programs.

Generally, applications involving the transcription of music to tape are thought of as more critical than applications such as news actualities, which deal only with speech. Relative perfection is a quality that is seldom appreciated in other than the music-transcription application. Part of the reason for this is that the "real-time" quality of broadcasting frequently prevents really top-drawer maintenance. Another contributing factor is that, in many cases, operating technicians do not have sufficient technical expertise to perform full alignments and fine spot "tweaks" of tape recorders. This condition makes it all the more important that routine full alignment be performed with scheduled frequency, thoroughness, and precision by a maintenance engineer; and further, that the tape equipment itself originally was purchased with long-term stability and precision in mind.

News and other field uses of tape machines require that ruggedness be more of a prime requisite than optimum aural performance. As a result, the machines chosen must be able to withstand substantial physical punishment, while still performing reliably.

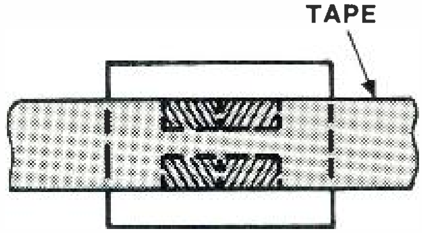
Production Studio Environment

Unfortunately, some station engineers hold the production studio, and the people assigned to do production work, at arm's length; almost as a sub-engineering species. The production area is truly creative, and requires great technical skill to properly perform. Frequently, disk jockeys are assigned to do production engineering. Some jocks have great production skills, and are able to create masterful production epics, whereas many jocks who do production work have miniscule technical ability. Frequently, the ego that is often considered necessary to on-air stardom, significantly hampers personal communication when the personality needs technical instruction. Similarly, the engineer may look down upon the jock's lack of skill, with the thought "If he think's he's so damn smart, let him figure it out himself."

A condition of cooperation and harmony must be established between these sometimes warring tribes before meaningful communication can take place.

It is appropriate that the engineer set the standards and maintain the production "tools" that the production technician needs to do the job. It is equally appropriate that the production technician learn from the engineer the correct use of the tools, and the proper application of the standards.

Front View of Correct Height Adjustment



Reproducing and Recording Standards

In any magnetic recording medium, if the recorded product is to be compatible among many machines, standards must be set for the mechanics, the magnetics, and the electronics.

Mechanical standards refer to physical aspects such as tape speed, track format, position of the tracks on the tape, tape width, tape guidance, head position, head azimuth, and head zenith. Clearly, many of these factors are set by machine manufacturers and industry associations. Parameters such as head position — including azimuth, zenith, and wrap — must be set precisely by the maintenance crew, against some standard.

Similarly, although the magnetic and physical properties of the head are controlled by the manufacturers, they require (in conjunction with the electronics) reference to some outside standard for precise adjustment, and a variety of gauges, tools, and test tapes are available for these purposes.

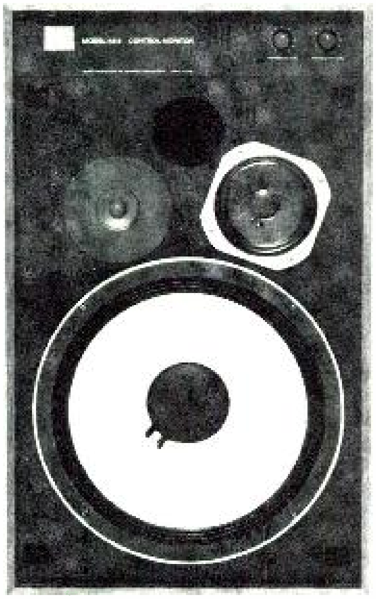
The electronics of contemporary tape machines are full of variables that must be standardized against outside reference, many times in conjuction with mechanical and magnetic factors.

The standardization procedure is relatively straightforward:

1) Standardize the reproduce head mechanically against an outside standard (such as gauges, tools and test tapes).

2) Standardize the reproduce electronics electrically against an outside standard (test tape, flux loop, etc.).

JBL Compact Monitors.



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For those that require a more flexible or compact monitor, we've created the 4411 and 4401. These loudspeakers incorporate our most advanced component and design technologies. Both the 4401 and 4411 utilize newly developed transducers arranged in a tight cluster to provide outstanding coherency of sound for close monitoring. This design also minimizes off-axis variations in the far field. Additionally, the 4411s are mirror imaged for improved stereo perspective.

For maximum flexibility, the continuously variable levels controls on the 4411 are calibrated for both a flat direct-field response and a rising axial response that produces a flatter power response. And for ease of adjustment, each of the monitors' level controls are baffle mounted. Finally, the low frequency loading has been optimized for flat response when the speakers are placed away from room surfaces. Because of this, the 4401 and 4411 may be console mounted without the loss of low frequency response typical of other designs.

For additional technical data and a complete demonstration of the 4312, 4401, or 4411, contact your local JBL Professional Products dealer. And discover the next generation of compact monitors. From the refined to the redefined.

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3) Standardize the record head electrically against the already standardized reproduce section.

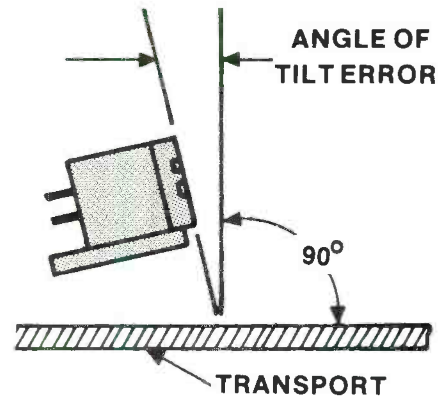
4) Standardize the record electronics against the already standardized reproduce section.

As simple as these four steps are, there are several different methods that can be used to accomplish the goal, some of which can yield anomalous results in different circumstances.

Reproduce Standardization

Reproduce standards are not only the first on the list, but the most critical, since later steps are standardized against this alignment. The most frequently used standard against which reproduce systems are calibrated is a Spot Tone Alignment Tape, manufactured by MRL, Ampex, STL, and others. This method is one that doesn't require a mind like Einstein to accomplish competently, and can yield very high degrees of uniformity and quality when accomplished with appropriate care.

The most common error committed when using test tapes as a reproduce standard is the making of one or more "copies" to save wear and tear on the original. This is unalterably *wrong*! Unless the technician making the copy has gone to extreme lengths to ensure that the machine on which the copy is to be recorded is in *exact* electrical and mechanical alignment, the resultant error probably will be perpetuated for years. It is certainly more advisable to buy multiple copies of the same alignment tapes on a regular basis — say at intervals not exceeding one year. Repeated use of alignment tapes will degrade their high-frequency accuracy, making it extremely important that any



Side View of Incorrect Zenith Adjustment

machine on which it is to be used has been thoroughly degaussed and cleaned *immediately* before an alignment tape is threaded.

Almost every professional tape machine maintenance manual contains a competent, step-by-step description of

KEEPING THE HARDWARE RUNNING
Hints and Tips for Troubleshooting Tape Machine Faults in the Production Studio
by Douglas B. Howland, Consultant Editor

A production studio engineer can do much to spot potential tape machine problems, and alert the engineering maintenance crew before a major out of tolerance condition occurs. Such tests include phase error, frequency response, level calibration verification, wow & flutter, and tape tensions. Some of these tests will require the use of one particular brand and type of recording tape — "the house standard" — since no two brands will perform identically. Each manufacturer produces tape with its own particular formulation. Many times the same brand and model tape will have slight variations between batches. As a result, any meaningful test should be conducted with one particular roll supplied by the engineering department. If you suspect a problem, go back to that reference tape to ensure that your current work reel is not at fault.

The production crew must be warned *not* to perform *any* calibrations or adjustments unless specifically authorized by the engineering department. In all cases where a problem seems to exist, other than authorized phase adjustments, the maintenance crew should be notified. This word of caution cannot be stressed too strongly, I feel, since any wrong adjustments eventually will show up in "on-air" material. At one major market station of which I was chief engineer, an announcer opened a sealed cover plate to adjust equalization controls in an attempt to make his poorly recorded tape — recorded at another station — sound "better." We discovered this tampering several days later. The announcer was fired shortly after.

. . . *continued across* —

HEAD ADJUSTMENT

To ensure the correct settings of tape head height and zenith, this simple hint will allow you to visually examine the effects of wear from a tape running over the newly installed heads. Simply "paint" the head with a soft-tip pen, thread a blank roll of tape, and play a short length. Upon careful removal of the tape, the ink rubbed from the head will reveal the wear pattern to be expected, and the effects of your height, zenith, and head rotation adjustments. Improper height setting will be revealed by the head gaps not being centered vertically in the tape path. This also could be due to incorrect guide adjustment. A trapezoid pattern indicated improper zenith setting. A symmetrical wear on each side of the gap will show tape wrap and head penetration.

how to accomplish reproduce alignment, reducing the necessity of repeating the steps outlined here.

As a general comment, it is important to realize that stereophonic and multitrack tape machines equipped with adjustable heads *must* be aligned mechanically for zero phase shift between channels at high frequencies. This alignment sometimes must be accomplished by trying a series of different azimuth settings, and choosing the one that provides the best overall phase compromise from 500 Hz to 20 kHz.

There are methods other than the use of spot tone alignment tape for aligning the reproduce section of tape machines. Standard or home-grown tapes containing tone sweeps or pink noise are frequently used by various engineers. In other cases, engineers induce a signal into the head from a "flux loop" fed from a signal generator to perform reproduce equalization alignment.

The sweep tone method can be displayed on a standard oscilloscope and, with a special film graticule calibration in dBs fixed to the faceplate, enables the high-frequency response from 500 Hz to 20 kHz to be read on the screen. Detractors of this method feel that the test is more a measure of the head than the electronics, and that it leads to anomolous results. This author has compared the readings from MRL spot-tone rapid-sweep tapes, and found the results to be in reasonable conformity. If a quick check of reproduce equalization as well as azimuth is needed, this method should be explored.

Several manufacturers of standard test tapes now offer pink-noise recordings, which can be used with third-octave spectrum analyzers to provide quick alignment of reproduce equalization. The cost of the analyzer makes this method expensive, but such an analyzer has other uses around the station, not the least of which is the record standardization process (more details below). While the flux-loop method can be highly precise — provided the operator is highly skilled and understands what he is doing — this method is a little complicated for routine maintenance.

The adjustment of high-frequency equalization is almost always well covered in the relevant equipment manuals. Low-frequency reproduce equalization must be adjusted with gap-width compensation factors in mind.

TENSION ADJUSTMENT

If you do not own a fish scale for setting reel motor tensions, simply add the correct amount of fish weights to the end of a length of string. Allow this end to hang free, while looping the other end around the reel on the motor under adjustment. Correct reel tension will occur when the weights remain stationary — usually 4 to 6 ounces with the machine in play mode.

KEEPING THE HARDWARE RUNNING . . . *continued —*

Test Equipment

The following items of test equipment will be required:

1) Audio Oscilloscope capable of Lissajous display (separate X-any Y-amp inputs).
2) Oscillator, minimum two-tone.

The following equipment is *recommended:*

1) Audio Oscilloscope as above.
2) Test Oscillator with pushbutton preset frequencies and sweep mode.
3) Pink-noise generator.
4) Octave level display.
5) Wow & Flutter meter.
6) Tension measuring device.

Such equipment should be installed so that the minimum number of knobs is exposed. All signals must appear on either a patch bay, or dedicated circuits. It is *not* the responsibility of production crews to fiddle with test leads, or calibrate the test equipment.

Performance Checks

A production engineer may check phase error and record-head azimuth on a stereo tape machine by recording a tone from the test oscillator on to both tracks, and observing the resultant Lissajous pattern on the oscilloscope by connecting the left output to the vertical display, and right to the horizontal (Figure 1). Start with a 1 kHz tone to check that the display is working properly. Now sweep, or switch, up to at least 10 kHz. The pattern should not deviate significantly from a straight 45-degree angle line. A 180-degree pattern reversal, with the display pointing in the opposite direction, will cause complete signal cancellation for the mono listener. Deviation between the zero- and 180-degree marks will produce attenuation of varying degree to the mono listener.

If a deviation is observed, the record head azimuth may be "tweaked." *Never* adjust the play head, since it serves as your reference, and has been calibrated by engineering maintenance against a master reference alignment tape. All other machines throughout the house will be calibrated against this standard. If you encounter outside tapes with phase errors, commercially manufactured devices are available for patching into the system to correct the playback phase/azimuth error. Any record azimuth adjustments also must be performed with these correction devices out of circuit.

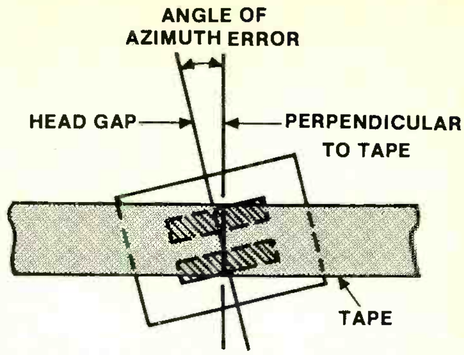
A rough check of the frequency response may be made by recording a 1 kHz tone while observing its playback level, and then switching to 10 kHz, taking note of the difference between the two levels. (Note: at 15 IPS this test may be done at normal recording levels; at

Record Standardization

After the tape machine has been made to conform to some external (and hopefully precision) reproduction standard, the record process is then aligned against the reproduce standard.

One, often misunderstood concept needs to be discussed here: fluxivity. Fluxivity is the intensity of magnetic flux recorded on the tape, and is expressed in nanoWebers per meter (nWb/m). Over the years, the level of fluxivity that can be impressed on to magnetic tape in the linear portion of the tape curve has steadily increased. In simple terms, this means that the track signal-to-noise ratio has steadily improved. The drawback has been that many early, and even some more recent, professional tape machines have been unable to impress very high signal levels (and complementary bias levels) on to tape without the machine electronics being overtaxed to the point of electronic distortion.

In 1964, the commonly used record fluxivity was 165 nWb/m, which usually corresponds to 3% third harmonic distortion at 700 Hz. By 1975, high-output, low-noise oxide formulations had allowed a fluxivity of approximately 250 nWb/m for the same conditions. And by 1980, further improvements in mastering tape had yielded a possible fluxivity of 350 nWb/m. Problems began, however, when users tried to adjust their older professional machines to produce these higher fluxivities, since both record amplifier headroom and bias oscillator output capabilities became limiting factors.



Front View of Incorrect Azimuth Adjustment

It is far better to choose a tape which can be effectively driven by your machine, and obtain the best performance possible on *that* tape, than to try to push the machine beyond its capabilities and obtain poor distortion performance. "Super Tape" may not be super for your machine!

The record head azimuth must be set as precisely as that of the playback head, using the latter for a reference. This is usually carried out by recording a high-frequency tone on both channels of a stereo machine, and adjusting azimuth until both channels peak on an output meter, then fine adjusting until they come into phase coincidence — usually by viewing a Lissajous figure on an X-Y oscilloscope. Alternatively, the same procedure can be done with pink noise, or a sweep oscillator for best compromise. On a single-channel machine, adjust the record head azimuth while recording 15 kHz for a peak in output level. Head zenith, height and wrap should be adjusted as stated in your machine's maintenance manual. Record head azimuth probably will have to be re-adjusted several times in the mechanical alignment process.

Record bias is a complicated setting, and there are many differing opinions regarding the most practical way to adjust a tape machine's bias. Due to differences in circuitry, bias oscillator frequency, tape and head characteristics, the setting will not be uniform among different models of machines. If a wave analyzer or the functional equivalent thereof is available, the most proper and consistent method would be to record a mid-frequency (say 700 Hz or 1kHz) tone on to the tape for which you are going to bias, and increasing bias from pot-down condition to the point where the third harmonic virtually disappears, as indicated by the wave analyzer monitoring the playback of the signal. Wave analyzers are expensive and not commonly seen in broadcast stations, however, so other methods must be considered.

KEEPING THE HARDWARE RUNNING . . . *continued —*

7½ IPS the test levels should be reduced by approximately 10 dB to prevent HF tape saturation.) Any difference greater than a few dB will indicate poor tape, dirty heads, improper tensions, or the need for electronic adjustments. The first step is to clean the heads. If that doesn't work, return to your reference tape for comparison. If the problem still persists, try another work tape. If that fails, call Engineering.

A more detailed check of frequency response can be made with an oscillator set to sweep mode. A slow sweep can be observed on a VU meter, while a fast sweep requires the use of an oscilloscope for display. Pink noise is a very good method when combined with a third-octave, real time display. An octave analyzer will allow you to see which bands of frequencies are affected by response deviation.

The combination of sweep tone and pink noise provided an excellent way of checking phase error, since all frequencies can be displayed via the Lissajous pattern on the scope. Checking phase by this method will prevent you from adjusting azimuth to a minor peak at high frequencies.

Level calibration may be compared by recording a 1 kHz tone, and switching the tape machine VU meter between input and reproduce. If there is a difference of more than a few dB, return to the reference tape before following Engineering's instructions. Again, different brands of tape have individual operating characteristics; your machine will be calibrated to one particular brand chosen as house standard.

Wow and flutter should be performed by the engineering department, since accurate measurements require an instrument designed for this purpose. Interpretation of the readings may be tricky for those unfamiliar with wow and flutter meters. If Engineering feels that they can train adequately, and that the need exists for production personnel to perform such checks, then a wow and flutter meter should be installed with the adjustments preset within range, and knobs unavailable for the curious. Speaking personally, I have always preferred to perform these measurements myself. Alternately, the main production engineer can be taught what the normal range of expected wow and flutter for a machine sounds like with a steady-state tone. If he "perceives" a deviation from the norm, then you can be notified.

Tape tensions are another measurement that should be made by the engineering department. Again it is possible to teach the main production engineer how to feel running tape, compare it against other machines, and to notify Engineering of "significant" deviations. There are devices available to give a direct-scale reading of running tape tensions, but these are delicate instruments that should be left in the hands of the engineering staff.

A sharp production engineer familiar with his or her surroundings, and provided with the proper test equipment and encouragement, can do much to alert the engineering maintenance crew of potential troubles before they reach the air-chain signal chain. ■

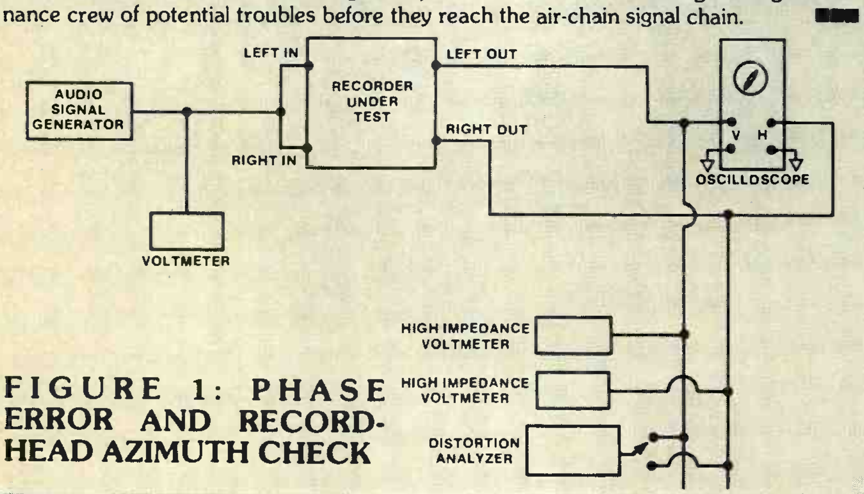


FIGURE 1: PHASE ERROR AND RECORD-HEAD AZIMUTH CHECK

Many tape machine maintenance manuals outline a practical method, normally referred to as overbiasing, which involves recording a mid-frequency tone, and adjusting bias while watching the reproduce VU meter. The bias is increased until the reproduce meter peaks, then increased further until the output level drops by from 1 to 2 dB. Harmonic and intermodulation distortion is then checked to be certain that it is satisfactory. Many other methods are used by engineers, and each must be carefully evaluated for validity.

After bias adjustment, it is necessary to adjust record equalization, since different bias adjustments cause different degrees of bias self-erasure of the high frequencies. In adjusting equalization, it is important not to try to over equalize the high-end response. At a tape speed of 7½ IPS, the frequency domain in which equalization difficulty occurs is usually from 10 to 15 kHz; at 15 IPS, the difficult region is usually from 15 to 20 kHz. Trying to push the HF equalization to make the highest frequency absolutely flat (with respect to the mid-frequency reference) will almost always result in a mid-range bump or accent (usually between 5 and 8 kHz). This can readily be seen if you are equalizing with a third-octave spectrum analyzer.

Usually, HF equalization is adjusted at 10 kHz to be flat with respect to the mid-frequency reference level, and the high-frequency extreme is allowed to ride where it naturally does. If the extreme frequency is greater than 2 dB down from mid-frequency reference, either you need to replace (or relap) your heads, or something is wrong electrically.

The last step in record alignment is to standardize the record (metered) level to the reproduce (metered) level, a process usually labeled "Record Cal." This simple process is often misunderstood, however. During the reproduce alignment process using a spot-tone test tape, a calibration tone at "reference level" is offered by the tape. Usually the tape will indicate the reference fluxivity. If you have adjusted your calibrated playback level so that this reference level reads 0 VU on the reproduce VU meter (and have locked the setting there), when you record a mid-frequency tone, and adjust the record level to reproduce at the same level, you will be recording a tone of exactly the same fluxivity. A pot usually labeled "Record Cal" is then adjusted to make the record VU meter agree with the reproduce meter. This is done by switching the meter between record and reproduce while setting the pot.

* * *

Reel-to-reel tape machines require frequent stem-to-stern alignment to assure top quality operation at all times. Although there are many different techniques that can be employed, the maintenance engineer must be certain of the validity of the technique he or she employs. The operator of the tape machine and the maintenance engineer who services the machine should develop an ongoing communication between themselves that allows service or operational problems to be communicated in one direction, and increased operational knowledge to be communicated in the other direction. In this way the audio production facility can make the most creative use of the recording and replay equipment at its disposal. ■■

DEMAGNETIZING/CLEANING

Although head gaps are extremely narrow — on the order of a couple thousand microinches — tape-oxide particles are even smaller — usually less than 20 microinches in diameter. Many of these particles can be scraped from the tape and lodge inside the head gap, remaining there even through the normal cleaning process. Sometimes oxide particles may be removed by soaking the gap with a cleaning solution, such as 99% Isopropyl alcohol, and simultaneously demagnitizing the head to draw them out of the gap space. To do this, first clean the record or replay head as best you can with a wet Q-tip. Then hold a second wet Q-tip on the gap for a few moments to ensure penetration of the cleaning solution. Turn on the demagnetizer's magnetic field, slowly bring it toward the head, and scrub the gap for buried particles. Draw the demagnetizer slowly away from the head at least three feet before turning it off.

New Products

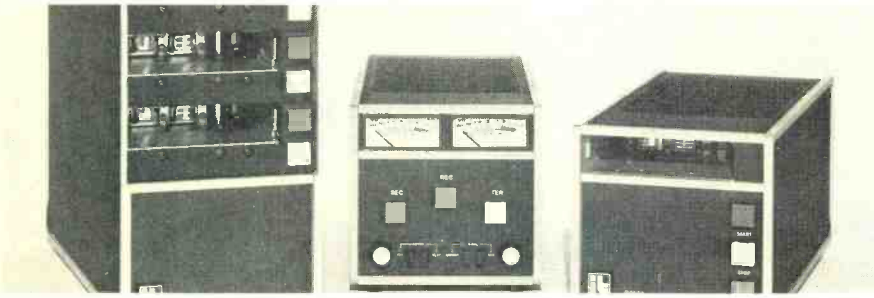
ITC/3M UNVEILS COMPACT DELTA SERIES CARTRIDGE MACHINES

The new Delta Series has been based on the well-known SP/RP/3D Series NAB cartridge machines, and now features a compact housing that occupies one-third the volume of the older units. By way of an example, the Delta One replay-only deck measures 5⅝ inches wide, enabling three units to fit across a standard 19-inch rack frame.

Additional, improved features on the Delta Series include:

- High-speed re-cue facility as standard.
- DC servomotor for capstan drive.
- Ceramic capstan shaft.
- Exclusive playback head design as featured in the top-of-the-line Series 99 decks.
- Full microprocessor control of transport functions, plus digital cue tone detection.
- Toroidal power transformer for decreased radiated hum field.
- High performance electronics, featuring NE6500 op-amplifiers.

All other RP/SP/3D Series features have been retained.



Prices will range from $1,715 for a mono Delta One replay deck, to $4,630 for a stereo Delta Three triple-stack.

ITC/3M
2425 SOUTH MAIN STREET
BLOOMINGTON, IL 61701
(309) 828-2381

For additional information circle #87

news

AEA TO HOST TWO-DAY "MINI CONVENTION"

According to Audio Engineering Associates, the purpose of the convention, which will be held at AEA's headquarters in Pasadena, California, April 22 thru 23, is to invite the audio community to participate in technical exchange and one-on-one demonstrations. Highlighting the event will be a presentation of the new Studer A810, a microprocessor-based, programmable, four-speed, SMPTE-capable, two-track recorder.

Exhibits also will include individual production suites — broadcast, audio/video sweetening, and recording — with new equipment from Lexicon and dbx. Quantum, Aphex, BTX, Otari, and Soundcraft equipment also will be featured.

For further details, contact Audio Engineering Associates, 1029 N. Allen Avenue, Pasadena, CA 91104. (213) 798-9127, or (213) 684-4461.

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MISCELLANEOUS

Studer Re-States the Art

With the new A810, Studer makes a quantum leap forward in audio recorder technology. Quite simply, it re-states the art of analog audio recording.

By combining traditional Swiss craftsmanship with the latest microprocessor control systems, Studer has engineered an ATR with unprecedented capabilities. All transport functions are totally microprocessor controlled, and all *four* tape speeds (3.75 to 30 ips) are front-panel selectable. The digital readout gives real time indication (+ or − in hrs, min, and sec) at all speeds, including vari-speed. A zero locate and one autolocate position are always at hand.

That's only the beginning. The A810 also provides three "soft keys" which may be user programmed for a variety of operating features. It's your choice. Three more locate positions. Start locate. Pause. Fader start. Tape dump. Remote ready. Time code enable. You can program your A810 for one specialized application, then re-program it later for another use.

There's more. Electronic alignment of audio parameters (bias, level, EQ) is accomplished via digital pad networks. (Trimpots have been eliminated.) After programming alignments into the A810's memory, you simply push a button to re-align when switching tape formulations.

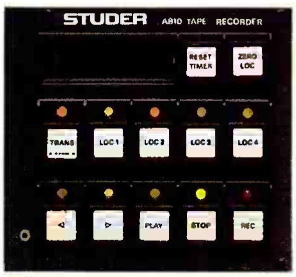
The A810 also introduces a new generation of audio electronics, with your choice of either transformerless or transformer-balanced in/out cards. Both offer advanced phase compensation circuits for unprecedented phase linearity. The new transport control servo system responds quickly, runs cool, and offers four spooling speeds.

Everything so far is standard. As an option, the A810 offers time-coincident SMPTE code on a center track between stereo audio channels. Separate time code heads ensure audio/code crosstalk rejection of better than 90 dB, while an internal digital delay automatically compensates for the time offset at all speeds. Code and audio always come out together, just like on your 4-track. Except you only pay for ¼" tape.

If you'd like computer control of all these functions, simply order the optional serial interface. It's compatible with RS232, RS422, and RS422-modified busses.

More features, standard and optional, are available. We suggest you contact your Studer representative for details. Granted, we've packed a lot into one small package, but ultimately you'll find that the Studer A810 is the most versatile, most practical, most *useable* ATR you can buy.

The Swiss wouldn't have it any other way.



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