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VOLUME 9 — NUMBER 3

INCLUDING: CONCERT SOUND REINFORCEMENT

Studio Design — *page 64*

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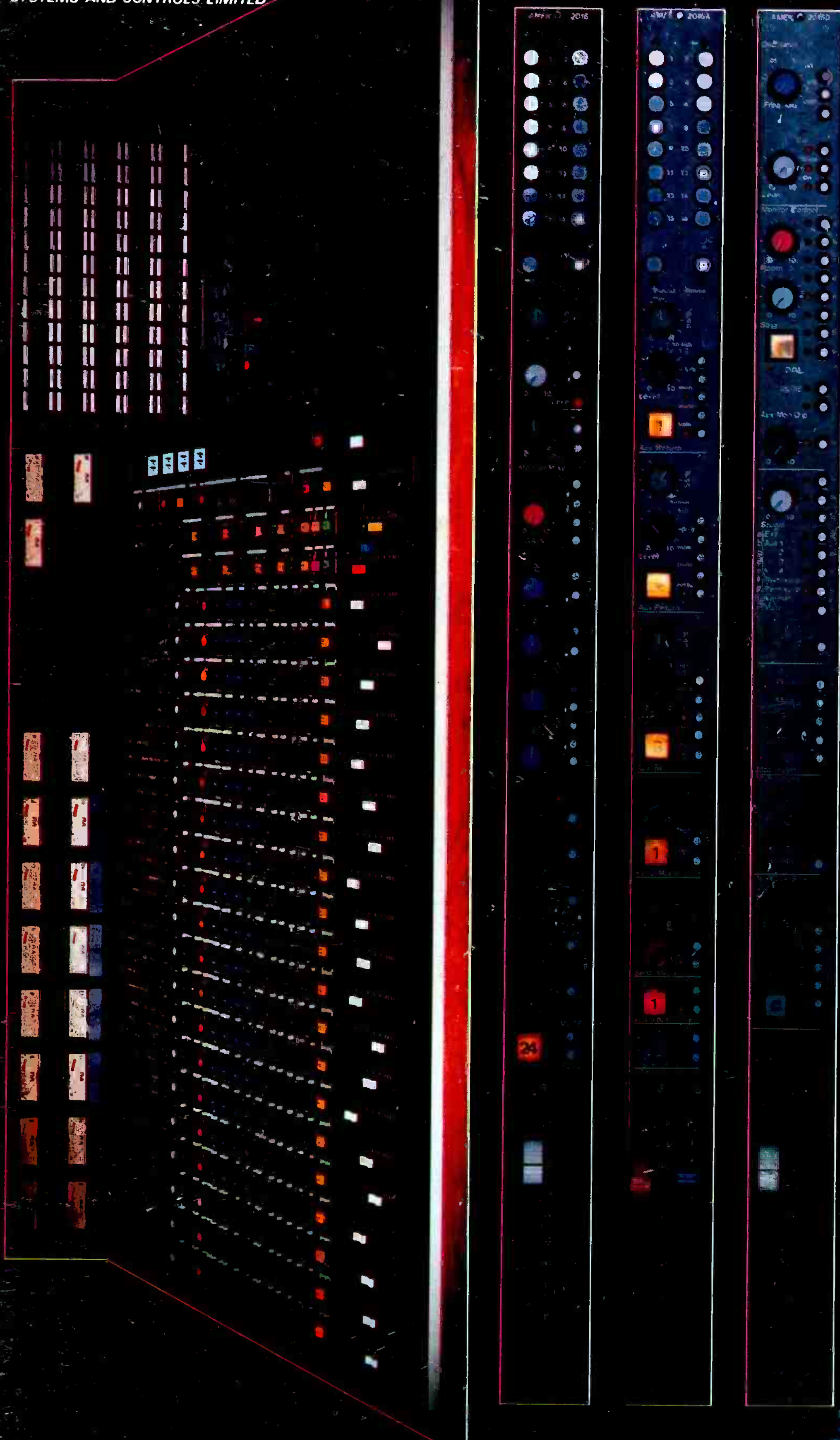

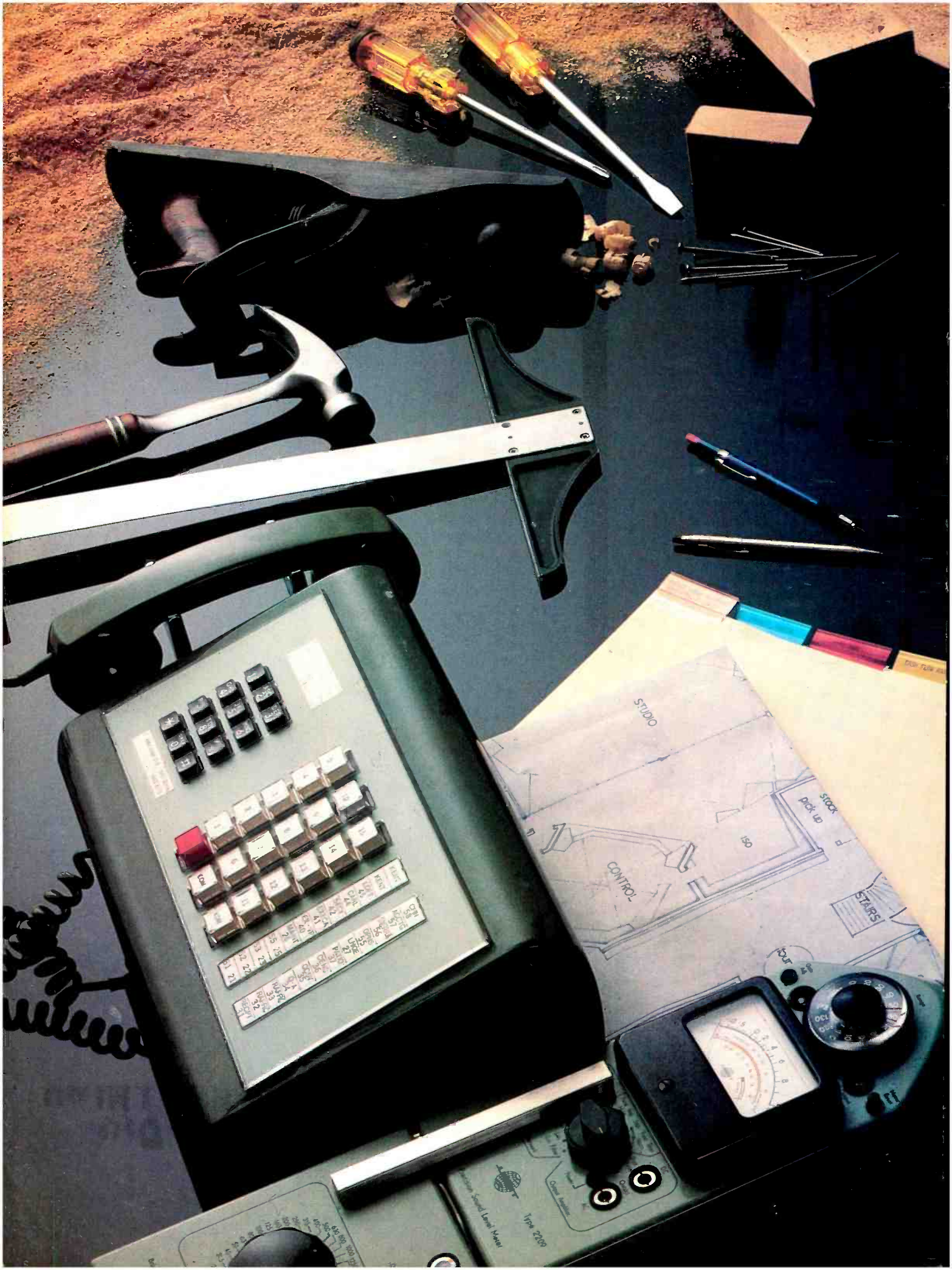
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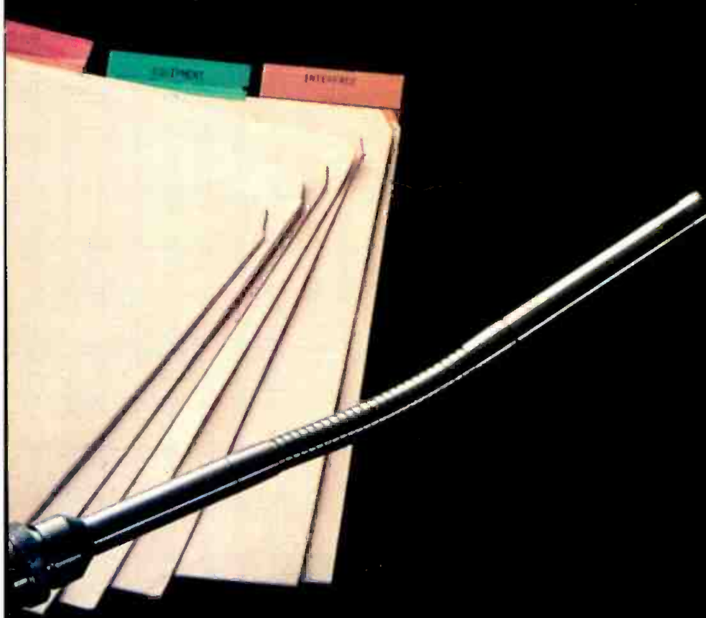

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— the magazine to exclusively serve the **Recording Studio** market . . . all those whose work involves the recording of commercially marketable sound.

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

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"**RECORDING** engineer/producer" is published six times a year by **RECORDING & BROADCASTING PUBLICATIONS**, 1850 Whitley Avenue, Hollywood, California 90028, and is sent to qualified recipients in the United States. One year (six issues) subscriptions for other than qualified individuals or companies may be purchased at the following rates:

United States (surface mail) . . $9.00
United States (air mail) . . . $17.00
Foreign (surface mail) . . . $9.50
Foreign (air mail) . . . $19.00

Controlled Circulation Postage paid at Los Angeles, California

Postmaster: Send form 3579 for address correction to:

RECORDING engineer/producer
P.O. Box 2449
Hollywood, California 90028
(213) 467-1111

JUNE 1978
VOLUME 9 — NUMBER 3

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the cover —
Photo by Robert Wolsch. Soundmixers Control Room "A", the Brill Building, Tin Pan Alley, New York. Studio complex designed for owner Harry Hirsch by Surgarloaf View's John Storyk. *(Story on page 64.)*

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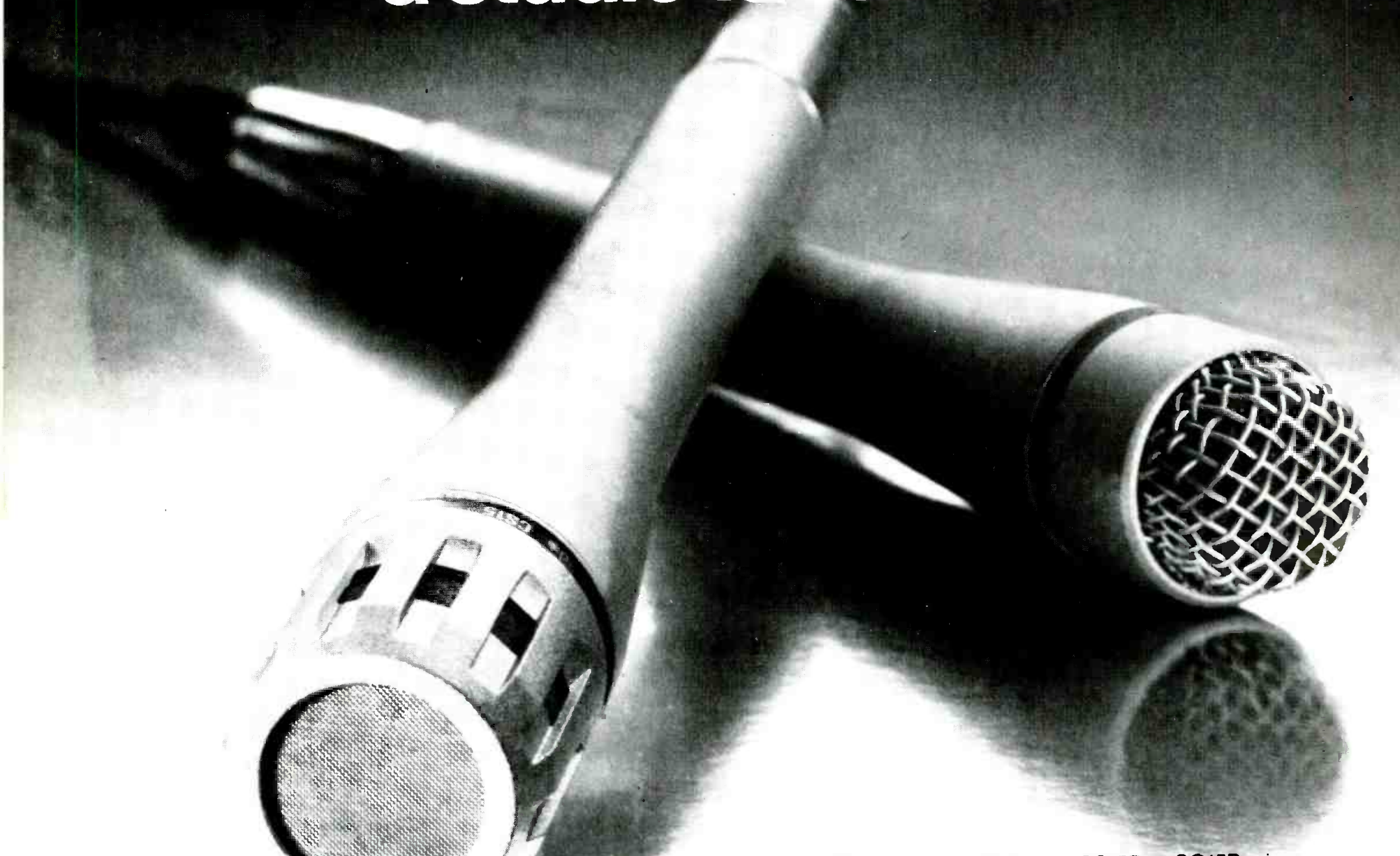

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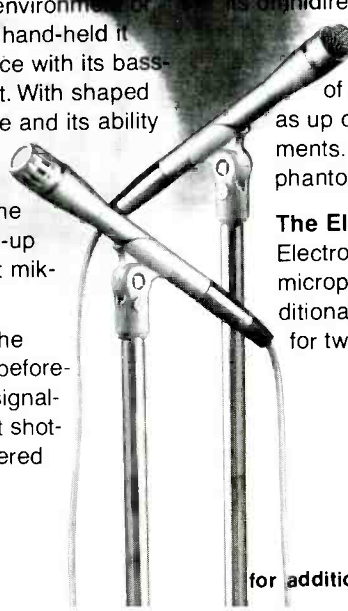

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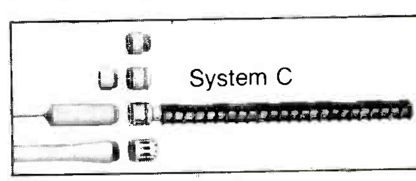

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Late News

AES DIGITAL STANDARDS COMMITTEE REPORT

Held in conjunction with the May Convention of the Audio Engineering Society in Los Angeles, the ***AES Digital Standards Committee*** attracted a record number of 54 attendees in addition to chairman **John McKnight.**

SAMPLING FREQUENCY STANDARDIZATION

The discussion of Standardization of Sampling Frequency was continued from previous meetings.

It was reported that **Soundstream** (Thomas Stockham), **Ampex** (Ed Engberg), and **EMI** (Reg Willard), had changed their respective equipment to the 50 kHz sampling frequency with satisfactory results.

In his paper "Notes on Digital Audio Topics", **Martin Willcocks** (Willcocks Research) suggested 52.5 kHz for professional and consumer machine compatibility.

Han Tenderloo (Polygram) reported on the progress of bandwidth reduction tests in Germany, in conjunction with PTT. Results are scheduled for release shortly.

Toshi Doi (Sony) in his paper "On the Selection of the Higher Sampling Frequency" suggests the use of 50.4 kHz as a sampling frequency compatible with 44.056 kHz low bandwidth rotary-head type video recorders. Support for 50.4 kHz came from **Jim Gibson** (RCA), who explained the compatibility between NTSC and PAL.

Both Ampex's Engberg and RCA's Gibson will submit additional pro and con statements on the 50.4 kHz sampling frequency prior to the next committee meeting.

A report by **Masahiro Fujimoto** (JVC), "Some Considerations of Sampling Frequencies for Digital Audio Systems" suggests that sampling frequencies should be different between professional disc and rotary head systems to prevent easy copying by pirates. This proposal was supported by the paper submitted by **Motokazu Ohkawa** (Toshiba). Discussion of this point suggested that the pirate would simply convert to analog for puroses of copying and the signal loss for this single conversion would be insignificant. It was felt that establishing non-compatible sampling frequencies only makes digital transfer more difficult for the legitimate user.

A subcommitee to collect, summarize and analyze discussions related to Sampling Frequencies, to be chaired by **Tom Hay** (MCI), was estalished. The desired purpose is to arrive at a minimum number of recommendations with a summary of advantages and disadvantages each recommendation entails. The report of this subcommittee is scheduled at the next Digital Standards Committee meeting.

SOURCE ENCODING

A paper by **Peter Sampson** (Systems Concepts) was presented on Incremental Floating Point Encoding.

Tom Stockham (Soundstream) and Ed Engberg (Ampex) both suggested that an optional sub-standard be considered in addition to the generally agreed upon principles on source encoding reached at the previous meeting in Atlanta. Stockham will prepare a statement for the committee on why such a sub-standard may be desireable.

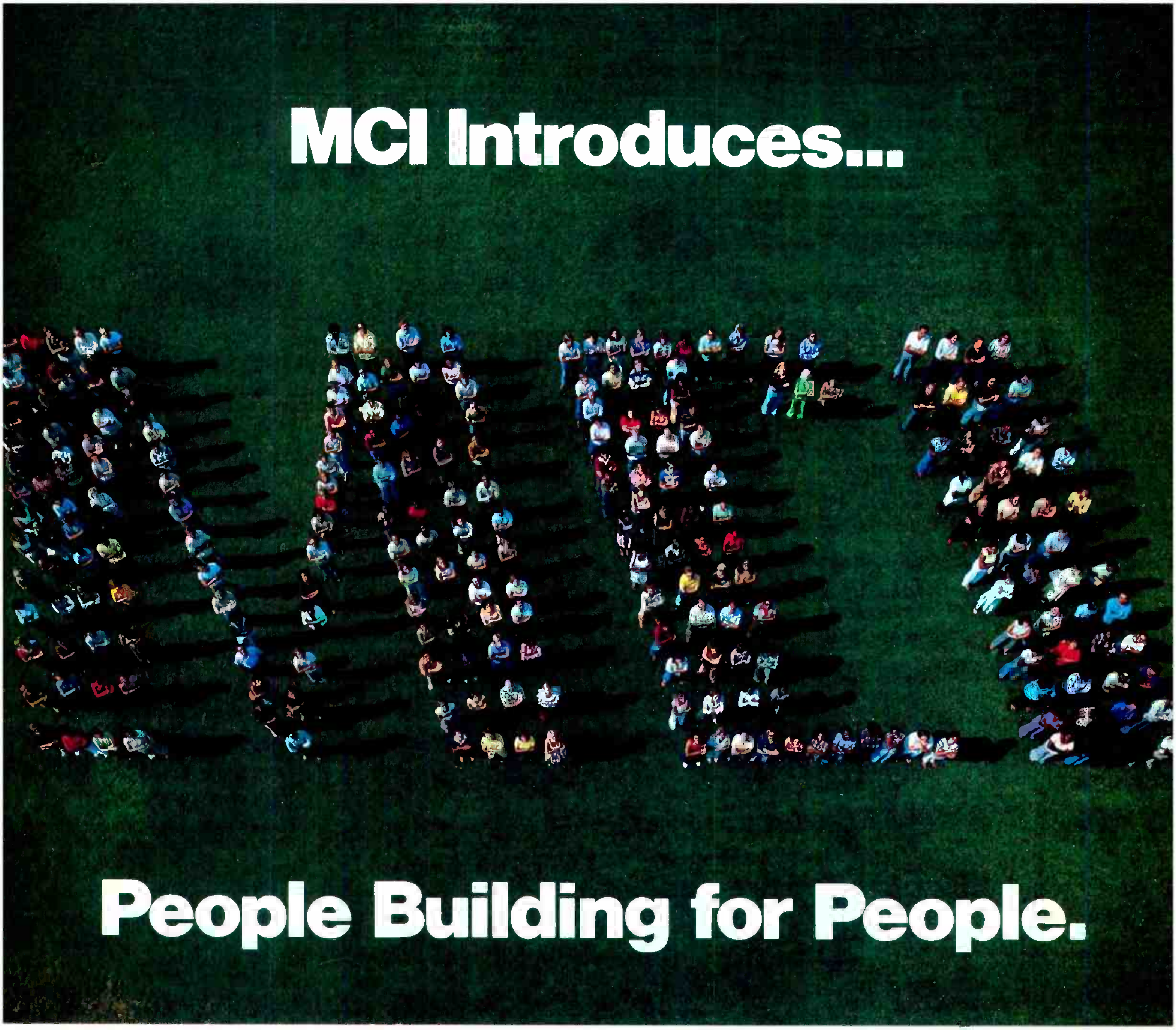
Henry Martin (Harrison Systems) reported on experimental work with a 20-bit internal sample representation.

"A motion to establish a sub-committee to study the requirements for digital source encoding and recommend to the committee at the next meeting on the suitability of 16-bit linear, or other possible codes that can defined" was defeated. This action did not constitute an official approval of the 16-bit linear proposal. Those interested in other source encoding should submit recommendations to Chairman McKnight no later than July 15, 1978, for discussion at the next meeting.

Eastman Kodak's **Rollie Zavada** exhibited a preliminary copy of an SMPTE document showing the signal path through film and television production. Tom Stockham will create and independent audio flow chart aimed at the professional market.

EDITING

A preliminary draft of a questionairre was discussed. **Curt Knoppel** (ITX)



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continued

will reword the document in terms understandable by the average audio editor. Results of this survey are required before a decision can be reached between block and interleave codes. Further work on an exact format hinges upon this decision.

CODE and MECHANICAL FORMATS

John McCracken (3M) presented a paper "A High Performance Digital Audio Recorder". **Ed Engberg** (Ampex) presented a verbal report on a new proposed format. Further discussion of error rates followed. **Norm Schwartz** (Heider/Filmways) was selected to head a sub-committee to assess needs of the users of longitudinal digital audio master recorders, to generate guidelines to ensure interchangeability of tape between machines of various manufacturers. The purpose of the sub-committee is not to define the on-tape format.

DIGITAL TERMINOLOGY

A committee was established to define appropriate terminology in the field of digital audio. A working group headed by Martin Willcocks will define the scope of this work and formulate definitions, as well as receive definitions submitted by other interested parties, to the remainder of the sub-committee.

The next regularly scheduled meeting of the Digital Audio Standards Committee will be on August 24, 1978 in St. Paul, Minnesota.

3-M ANNOUNCES DIGITAL SYSTEM ENHANCEMENTS AND REVISED PRICING

In announcing a revised pricing policy for their two-unit digital recording systems (a 32 track pre-mix recorder, and a two/four track mastering recorder) marketing director Robert Brown explained: "The company had originally projected a purchase price for the system to be under $150,000; that figure still accurately represents the value of the system.

"However, because we're taking the leading role in introducing a new technology to the studios, and further refinements are sure to evolve, we feel we must share in that responsibility. This can be most effectively accomplished through a lease-rental arrangement."

The financial arrangement announced at the May AES convention includes a reservation/installation fee and monthly rental of $4,000, plus a usage fee of $4 an hour.

Among the enhancements announced are an improvement in Latency Time — in getting into and out of the recording mode — has been reduced to approximately two milliseconds. It had been as much as ten times that, because all key functions were multiplexed. This has been refined by removing the record, stop and play functions and making them independent of the common multiplex system. The improved time will aid mixing and dubbing operations, according to 3M.

Record-mode LEDs are now used for each of the 32 channels. On the original prototype, individual *arming* lights were employed with a common *master* record-mode indicator. The use of individual lights for the record-mode will permit operators to tell at a glance which channels are *armed* and which are actually in the recording mode. This is of particular benefit when in the sync-record mode.

Half-inch tape will be used on the four-channel recorder, rather than the quarter-inch. This change is being made in the interests of more reliable and convenient handling with 14 inch reels, minimizing the potential for any physical damage to the tape.

"There was no track-width problem originally", says 3M, "there could have been eight tracks on the quarter inch tape. However, since the present configuration puts only four tracks on half-inch tape, the tracks are made somewhat wider." 3M points out that there was good reliability even on the quarter-inch tape, "but making the tracks wider affords the opportunity for increased reliability."

In commenting on the progress of their digital program Dr. Marshall Hatfield, general manager of 3M Company's Mincom Division said, "Prospective purchasers have outnumbered the systems to be initially available.

"We're committed to the originally announced schedule of three units toward the end of 1978, with continued production starting at our Camarillo, California plant early in 1979.

"3M Company is entering the recording stream at the studio, which will provide improved masters that result immediately in improved stereo

continued overleaf ---

3M
continued

discs and pre-recorded tapes. This will be true, despite the fact that the final production and home reproduction is by analog equipment," Hatfield said.

"Within a very few years, however, we're convinced that there will be practical, compatible digital disc and tape systems for the home. We feel that today's introduction of our complete mastering system gives impetus to the development of such home equipment.

"A revolution in home entertainment systems is imminent," Hatfield continued, "and the digital studio master is the key starting point. What everyone has accepted as high quality up until now will no longer be adequate. Once we marvelled at the quality of '78' records, but then we heard 33 LPs with their dramatic increase in fidelity and decrease in background noise. The contrast between analog and digital is as dramatic."

THE SOUNDSTREAM DIGITAL SYSTEM
-in use -
by Bruce C. Rothaar

Digital audio now has a foothold in the commercial marketplace, as evidenced by the increasing number of record companies contracting for digital recordings, and the increasing number of companies developing digital audio equipment (such as audio delays, mixers, and recorders). The first records to be recorded and edited digitally and mastered from digital tape are already being sold in audiophile record stores.

This article compares record production using current digital techniques with production using conventional equipment. Observations are those of Soundstream, Inc., which for the past two years has made digital audio recordings in the field and mastered records from digitally edited material. The Soundstream effort has been to utilize digital technology to bring to the recording industry the highest possible quality of audio recording.

The Digital Audio Process

In a digital recorder, signals applied to each input are sampled at regular intervals and a number representing the amplitude of the waveform at each sample point is written, as a group of bits, on a recording medium (such as tape). During playback, the numbers are read from the medium and converted back to the voltage levels which they represent, producing a continuous signal at the output of the recorder.

The number of bits used for encoding the sample values, and the type of encoding, determine the dynamic range of the converter as well as the signal-to-noise ratio. For linear encoding the noise level (quantization noise) is constant for all signal levels. Soundstream's recorders use 16 bit linear encoders, which provide a dynamic range of 90 dB and a S/N for peak level signals of 90 dB. Floating-point or logarithmic encoders will produce noise levels which follow the signal level, yielding a much lower S/N at peak levels than with linear encoding. This effect is undesirable in audiophile recordings.

Close inspection of the digital audio process reveals the possibility of other types of signal distortions. These distortions, introduced by some analog-to-digital and digital-to-analog converters, include harmonic and anharmonic distortions and T. I. M. Soundstream has minimized these distortions through a careful theoretical approach and sonic evaluation.

Features of Digital Recording

Since there are no low frequency roll-off elements in Soundstream's recorders, frequencies as low as 0 Hz (DC) can be recorded. Waveforms are sampled 50,000 times per second, ensuring a flat frequency response to beyond 20 kHz. And since numbers are encoded onto tape as saturated pulses, they are impervious to the traditional degradations of analog tape, eliminating audio print-through and permitting long-term archiving without signal degradation. Wow and flutter caused by the tape drive are completely corrected by loading the numbers, as they come off the tape, into a small memory and unloading them at the precise rate of a crystal oscillator.

Copies of digital material are indistinguishable from the original because the copying process is like transcribing text from one sheet of paper to another. And since the numbers representing a signal have the same form as numbers found in other digital systems, they can be processed mathematically to achieve any of the effects of analog signal processing. Other effects, not possible in analog processing, can also be realized. (Imagine a graphic equalizer with 1,000 frequency bands, each of which can be adjusted by up to 90 dB. This type of digital filtering is common practice.)

The Recording Session

A digital recorder produces extremely faithful recordings which better reflect the quality of the performance. The demand on the studio engineers to provide an excellent feed to the recorder is therefore greater than for an analog recording session. Placement of microphones need not change, although the choice of microphones may. Most microphones distort peak level signals when used over a 90 dB dynamic range. Instrumentation microphones are very accurate and have been used successfully in several digital recording sessions.

Upon arrival at a studio, a Soundstream recorder is qualified for proper operation by monitoring the output with a distortion analyzer for a variety

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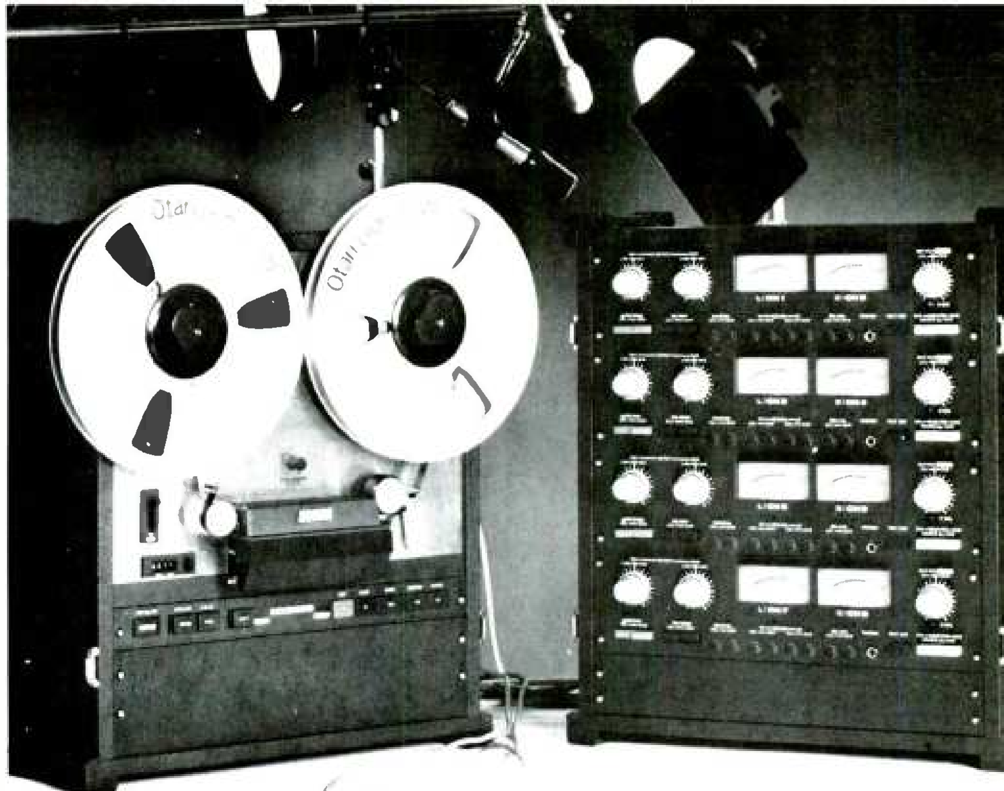

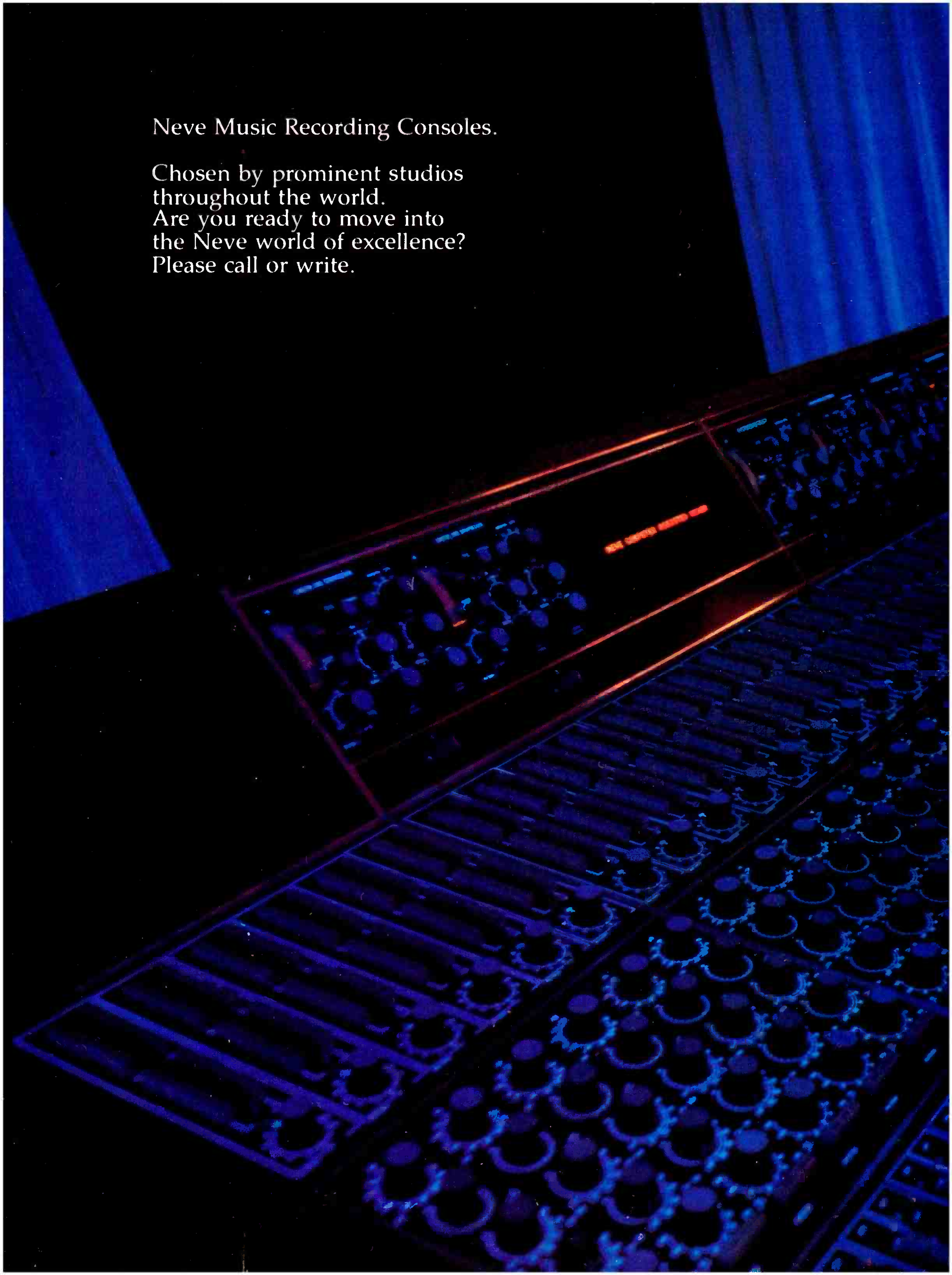
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engineer ART STEWART and the MOTOWN sound

text and photos by
Howard Cummings

Engineer-Producer Art Stewart first became interested in audio through Motown engineer Cal Harris (presently working with The Commodores). Cal took him into Motown-Detroit, put up a tape and Art watched, intrigued, as Cal began playing with the sound, "changing and mixing it. I thought it was great. I knew this was for me since I had a little electrical background working at GM during the day and going to night school playing with resistors and circuits."

We started the interview by talking about his old stomping grounds, *Hitsville,* the Motown studio in Detroit famous for the hits of Stevie Wonder, The Temptations, Four Tops, Supremes, and many others.

Art Stewart: *Hitsville* was a remodeled house on West Grand Boulevard, and it was basically set-up the same way for every session. The piano player was in the same place, the drummer, etc., with the same mikes and placement. Motown was really into the KM86s and the engineers would follow suit.

Howard Cummings: Because it was a pretty consistent sound, like the drums for example.

Art Stewart: Yes, and the musicians and the engineers were really tuned into the sounds. Today, we're more into improvising. The engineers use a vast array of mikes along with different musicians. In those days it was James Jamison on bass, two drummers on the same session at the same time — Ural Jones and "Pistol", Jack Ashford on percussion, Eddie "Bongos" on bongos and congas, Earl Van Dyke on keyboards, Eddie Willis, Robert White, Dennis Coffey, and Joe Messina on guitars — at one time they'd use all of or a combination thereof. These guys would play on all the sessions. The console was an in-house custom board and 604Es.

Howard Cummings: What would you attribute the "Motown Sound" to?

Art Stewart: With that particular sound in Detroit, it was a combination of the engineers, the room, and the musicians — maybe more-so the musicians.

Howard Cummings: What did the engineers do to "shape" that sound?

Art Stewart: I think there was a very good rapport between the engineers and the musicians. If a guitar was playing too loud, he'd turn down and accommodate you, so that made the job easier. If you can get the musician to work with you, that's a big part of getting the job done, instead of having to deal with a lot of loud amps and leakage. If a drummer were subconsciously playing too soft, then too loud, you could bring it to his attention so you wouldn't have to continuously move your sliders up and down. It cuts down on the herky-jerky type of engineering. The musicians were great in helping out.

The engineers were very attentive in mike placement. There was good communication between engineers, musicians, and producers.

Howard Cummings: What about the room itself?

Art Stewart: I think the room itself was acoustically phenomenal in that the ceiling was half falling, there were holes in the walls. They had a few acoustic tiles up, but I think it was done to cover the holes! (Laughter) I don't think they were there to treat the room.

HC: Was it rectangular or circular?

AS: Basically it was a house with a step-down livingroom, no carpet, no drapes, and some acoustic tile here and there. The size was about 30-35'. One of the nice things was they had

continued overleaf . . .


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three isolation rooms towards the back of the studio. These cubicles could be used to set up your vibes or other acoustic instruments.

That studio was so influential though. It even attracted British guitarist Jeff Beck and Mickie Most to do some Motown tunes with the Motown rhythm section.

I think one of my first sessions was INDIANA WANTS ME, by R. Dean Taylor. Don Gooch was his engineer at the time and I was filling-in on some vocal parts. One of the first big things I worked on was with Stevie Wonder. That was engineered with Orson Lewis and was Stevie's WHERE I'M COMING FROM — going through his new phase. New for him and new for me. It was amazing to see the guy go out and lay-down drum parts and then lay-down some keyboard parts and watch it build.

HC: Did he use click tracks?
AS: He just kept time himself. Stevie's phenomenal in that. On occasion he would put down the guide track, then go back and put something else on it.

HC: What as a guide track?
AS: If the time got a little shaky in spots, but was goog enough to work with, he could use the keyboards to lay down a solid track, then go back and re-do the drums. Sometimes he would even use the keyboards as the original guide track.

HC: How did he feel about how his keyboards would sound — was he into more of a "feel" or more technically oriented?
AS: Stevie is more of a "feel" man. If it had a right feel to his ears, he would let it stand.

HC: How did you go about accommodating him in that area?
AS: For keyboards, we'd go direct. If we used acoustic piano, we'd use a couple of 86s with tight miking down near the sounding board, and we wouldn't have to pad the piano because it was a one-man band so to speak. I'd just leave the lid up and get a better sound that way.

HC: And his vocals?
AS: Since Motown had about a zillion 86s, 67s, 84s, and 87s, 635As and 666s, we used those on basically everything — probably 87s or 67s.

HC: Since he usually became pretty animated on his vocals and would weave back and forth while singing, what did you do about his drifting off-mike?
AS: A couple of the engineers would set up two mikes since he moved left to right, but when I recorded him, I used one mike. If you'd tell him about it and make him cognizant of it, he'd remember to stay in one place.

Another interesting thing; if another person were producing him and it wasn't Stevie's song, Stevie would sing one line of the song, and while he was singing, the producer would give him the next line over the cue-feed. For example, if he were singing "My Bonnie lies over the ocean", the producer would feed him "My Bonnie lies over the sea" — into the cue system. We'd run down the song in that manner — we'd fast-phrase the melody. But naturally if it was his song, he'd remember what was going on.

Once when we were working on SIGNED, SEALED AND DELIVERED in Detroit, he didn't sing the line exactly as it should have been. After looking at the lyric sheet, I said, "You didn't sing the line right." He told me, "It sang well, it felt good, so let's leave it like that." He's a bit of a technician and he wants it right, but if it felt good it's unimportant that the line wasn't exactly the same.

HC: By the time Stevie Wonder had turned 21, he had renegotiated his contract and started doing what he wanted to do.
AS: That was WHERE I'M COMING FROM. I noticed that distinct change and I think that album was ahead of its time. Personally, I think that when he turned that album into Motown, they didn't realize what they had.

HC: No, because it was a complete about-face. They had "Little Stevie

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ART STEWART / MOTOWN

Wonder", harmonica kid, FINGERTIPS, etc., and all of a sudden he's into Mini-Moogs, synthesizers, Polys, and they had to figure out "How are we going to market this new one?"

AS: I think there was a lot of thumb-twiddling at the time. "What's our next move? Why isn't he singing FOR ONCE IN MY LIFE?" That was quite an experience.

Soon afterward I did a lot of work with Johnny Bristol, Frank Wilson, and Norman Whitfield's PSYCHEDELIC SHACK, CLOUD NINE, RUNAWAY CHILD.

HC: Now with the Temps, you're dealing a lot with harmony vocals.

AS: Norman Whitfield is a master craftsman. My job was to record it properly and working with the Temptations, it was great to see this guy take five voices and have it come out right. It wasn't always easy.

HC: And I assume you would not do everything live — you had already done your rhythm tracks beforehand?

AS: Right, that was more or less Motown procedure — to have the rhythm tracks first, then either go to vocals, then strings, or strings *then* vocals. Normally when you did rhythm, you might do lead vocals to help the track along.

HC: What about the story that Motown used to cut different tracks and then bring in different artists to see how they'd fit with the track. For example, I had heard that the arrangement for Marvin Gaye's GRAPEVINE was in the can a year before it was released, waiting for another vocalist.

AS: If Norman Whitfield believed in a tune, he'd cut it on 10 artists, he was a master at that — just check his track record. All of his artists made his tunes.

HC: But in the fall of '67, Gladys Knight & The Pips had a Top 10 record with GRAPEVINE. A year after that, Marvin Gaye released his *version, and had the biggest single Motown ever had!*

AS: Norman was great at doing that. As a matter of fact, Marvin's had been an album track and a DJ picked up on it and it took off.

But if a track was cut, and for some reason it didn't come off, you could take the same track, remove the old artist, and once you put on a new artist that would negate the old artist's financial obligation. It's still done that way today.

HC: But back to the Temps. How would you arrange them in the studio?

AS: At that time, you would do the background parts, then the lead, or vice-versa, not all at the same time. That's how they recorded in Detroit.

Motown had the greatest producers in the world. They could do it that way. Sometimes they'd make up the background parts right in the studio — they were inspired right there. Norman would say, "I want you to do do-wah do-wah, and I want you to do wah-wah, and I want you to do diddy-wah, diddy-wah." So now you know if you're the tenor and you're the bass, those are the notes you sang. "You back up, you come in until you get the balance."

HC: If you had three or four guys on harmony vocals, would you use multiple mikes or four guys on one mike?

AS: Sometimes you'd use *dual* miking, very seldom multiple, again back to the 67s and 87s. 666s were sometimes used for lead vocals. I have used the 666s here in L.A. They're very durable.

HC: The same technique for the Four Tops?

AS: Levi Stubbs was amazing. If you dubbed him in today and would go home and listen, then found out it wasn't as good as it could be, you could tell Levi to do it over, *he* could come in and match his voice! You wouldn't have to do anything. He'd listen, match it and you couldn't tell the difference between yesterday and today. He was phenomenal.

continued on page 30

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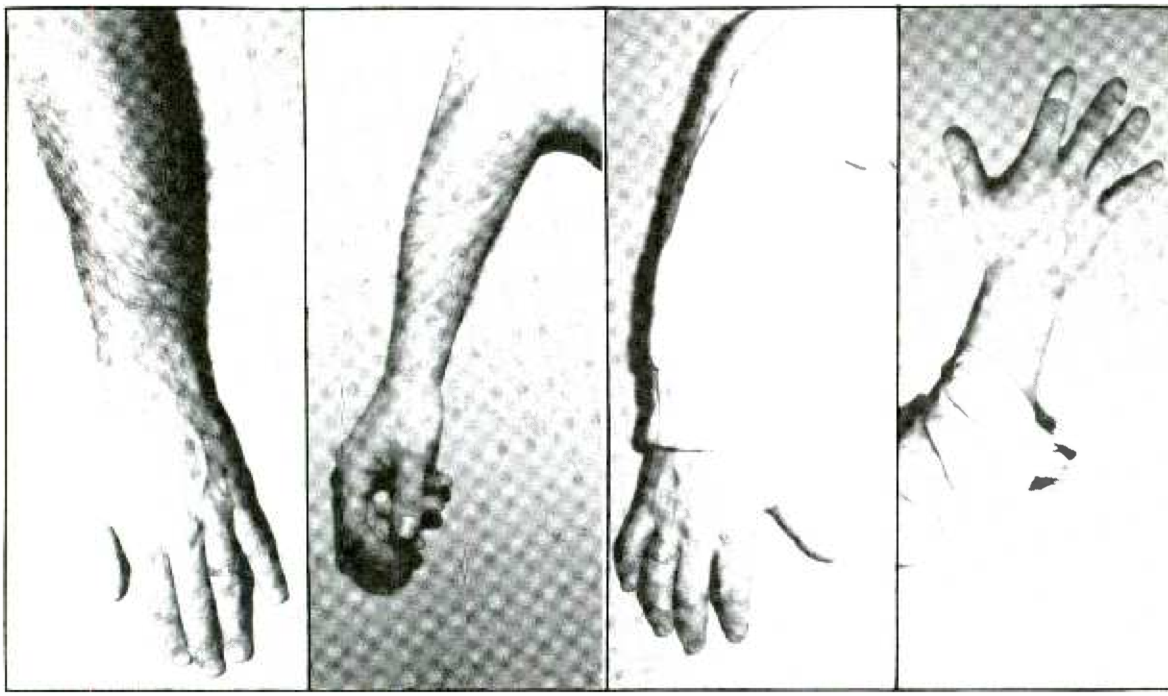

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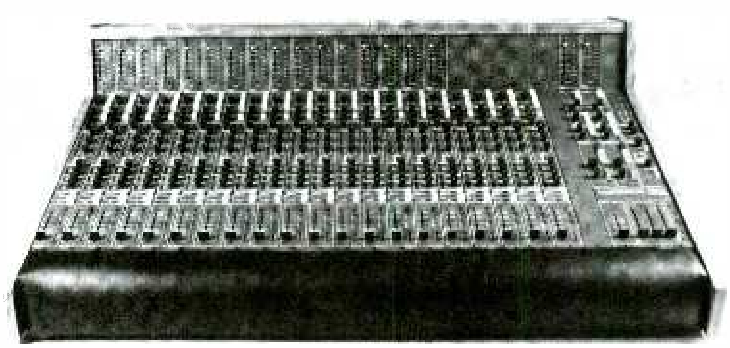

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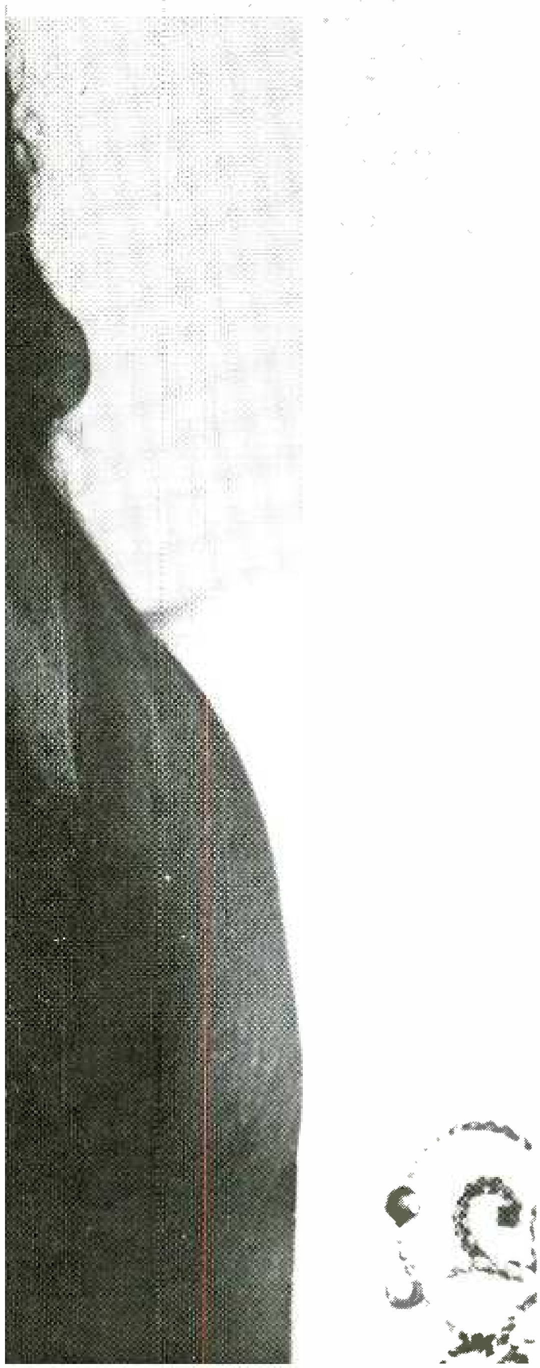
"FOR THE FIRST TIME RECORD PEOPLE AND MOVIE PEOPLE HAVE REALLY PULLED IT TOGETHER."

Quincy Jones

It's not surprising that Quincy Jones sometimes feels like he was born in a studio. He's performed on, composed for, or produced over a thousand albums. Right now he's finishing his first musical, Sidney Lumet's version of The Wizard of Oz, The Wiz, starring Diana Ross.

While Quincy is one Jones that's impossible to keep up with, we were able to catch him briefly to find out his views on the current recording scene, his latest work, and "Scotch" 250 Mastering Tape.



The only thing Dizzy Gillespie, Andy Williams, Peggy Lee, and Ringo Starr have in common is that they've all worked with you. How can you work in so many musical styles?

"I don't get hung up in any bags. When I was studying in Paris, a teacher told me once, there were only twelve notes, so you should find out what everybody's done with them, because they're the same twelve notes that Palestrina was scuffling with. So I can live with the best of all different areas. I like that, you know. The menu is broad, man – eat everything."

There are a lot of movie scores that have turned into some pretty hot albums lately, Saturday Night Fever, for example...

"You know why I think it's happening? It's just a guess... for the first time record people and film people are basically the same people and they've really pulled it together.

"Of all the films I did, the thing that bugged me the most was that we'd be in the studio and the music would boom down at you, and when you got to the theatre it was almost like a rumor, all the bottom end and the top end falls off. Then Dolby came along and they got *A Star is Born, Star Wars, Close Encounters,* and *Saturday Night Fever.*

"Those are successful record-wise because for the first time people actually hear the music in the track, really hear it. We've got a new kind of sound system now with Dolby. Emotionally it hits you from a place you're not even aware of."

Is it technically harder to achieve what you want in a musical as opposed to doing a score for a dramatic film?

"Oh yeah, in *The Wiz* we've got choral things that go up to 80 and 120 voices, so to get a good lip sync we decided to use just two voices for guide tracks, almost like a Polaroid. After their mouths are moving in the right way, then we sit down and put the sweetening on the dance and singing numbers."

So the music is composed simultaneously with the filming?

"They've been sending me out dailies on videotape from New York because the color really turns me on. You get it at 2 o'clock in the morning and look at the reel about ten times. You have to eat it. That's the best homework you can do for a film."

You're a big user of "Scotch" 250. Do you find that it has a clean sound? That's one of the things we've been selling the tape on.

"That's right.

"It's like with film stock, you know. When you've got 800 people out there on a set, I don't care what happens on that performance, if it isn't recorded on camera, it's all over. And it's the same in the recording studio: everything else is superfluous.

"No matter how great a song we get, or performance or balance or anything else, if that same thing isn't reproduced and captured on that tape, nothing we do means a thing.

"That's why we stay with 'Scotch.'"

"SCOTCH" 250 MASTERING TAPE.

The tape the masters use.

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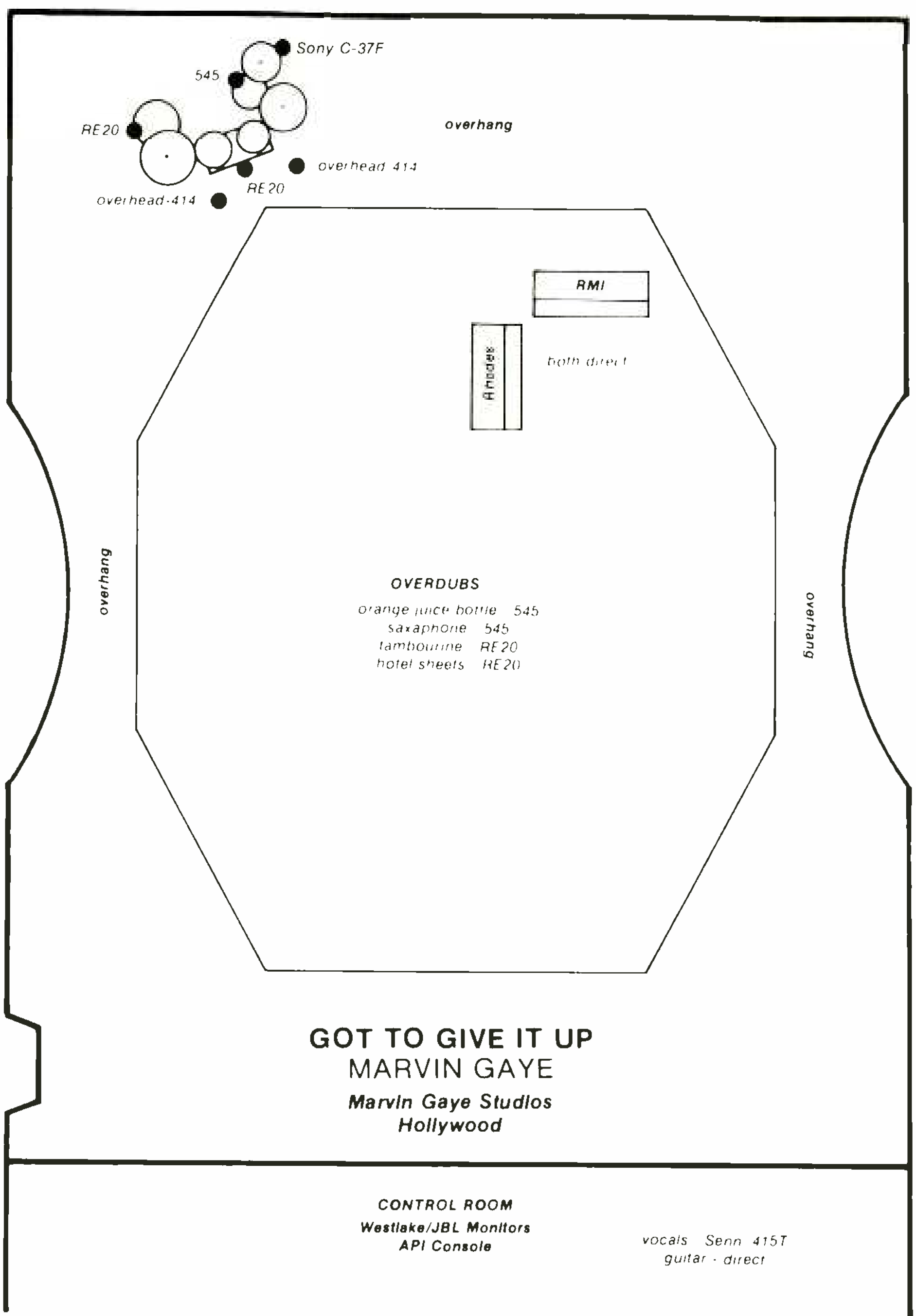
ART STEWART // MOTOWN

HC: And sometimes it's hard to do with the ambience, EQ, and level, not to mention the guy's voice changing from the time of day.
AS: I have a standard way I record, I use it for everyone. I don't EQ microphones, I really don't. If anything I try to select the right mike. The only way I'll EQ a mike is if the producer insists. If he makes a comment that it doesn't sound right, I'll say, "Well, hold on. Let me change the mike, you might like it better that way." I'd rather do it flat. If you were on staff and you were coming in behind me, you'd know Art Stewart records it flat. You could put up the same mike and take it from there.

Going back to Levi Stubbs; other artist's voices change from day to day, but with Levi, it didn't make any difference. He'd do the work for you — great control.



HC: Did you work on LOVE MUSIC?
AS: No. In 1970, I moved to L.A.

HC: I thought they might have recorded that out here.
AS: When I left Detroit in 1970, I quit Motown-Detroit, but rejoined Motown-L.A. after a year. Larry Miles, who worked on BOOGIE FEVER with Freddie Perren, came out with me on the flip of a coin. I quit on a Wednesday and arrived in L.A. on a Thursday.

Larry rejoined Motown-L.A. at a place called Sound Factory West and they were working him to death over there! So Larry asked me to approach them after a year and I was hired as an engineer, not as an assistant.

In the year I was out here, I was doing free-lance sound for the major networks. One of the first things I did was to go out to the Western White House for President Nixon. The big-time right away!

HC: (Laughter) Were you the one who taught him how to record?
AS: Yeah! (Laughter) That was quite an experience.

At Sound Factory West, I did some work on the Jackson 5 with Freddie Perren and Fonce Mizell of The Corporation.

HC: Who were the guys functioning under the name of "The Corporation"?
AS: Berry Gordy, Jr., Deke Richards, Fonce Mizell, and Freddie Perren.

Then we moved into the new Motown and started working on movie scores like HELL UP IN HARLEM, THE MACK, and FOXEY BROWN with Willie Hutch.

I found out engineers are very valuable if they can communicate with the producer or make the job easier, especially since producing these days is into punching-in a breath or a sigh and the producer wants to save the line just before the sigh. You have to be very attentive. If you erase the sigh, you ruin a whole year's worth of work. (Laughter) But that silly garbage is ridiculous.

HC: To try to punch in a small line?
AS: Yes.

...five years ago, I've done all of Shaun Cassidy and Leif Garrett on it, most of Donny and Marie, plus Al Martino, Sammy Davis, Debby Boone, the Supremes and others."

"After 20 singles and 12 albums on this console, 25 of them gold and platinum, I guess you could say my Auditronics 501 (serial number 0002) is a real money-maker. It does what I want as well as when it was new, and I'm still cutting on it today."

Independent producer Michael Lloyd is one of over 300 satisfied Auditronics console users. If you'd like to learn some of what they know about Auditronics console quality and reliability, circle reader service number or write to

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for additional information circle no. 18

HC: What would you suggest as an alternative?
AS: To do the whole line up until the end. But I've done a lot of recording like that and in working, you become a victim of your dislikings. Even in my productions, I hate it, but you may have worn yourself thin trying to get a certain performance. If a performer's run out of gas, you take what you can get. But just for feelings' sake, I think it's better to do the entire line instead of punching in and out a lot.

HC: And this album you're working on now with Marvin Gaye . . .
AS: It's tentatively titled HERE MY DEAR.

HC: What stands out for you on this album?
AS: I think the songs themselves, and the concept. The concept is dealing with emotions; the frustrations of love, marriage, ups, downs, divorce, and being centered around a broken affair. Then there's a song titled ANGER about a guy who is angry and not really wanting to be. There's one called LOVE, where everybody and everything needs love. It could conceivably become a standard.

These are all tentative because he writes sitting at the console. GOT TO GIVE IT UP was done sitting at the console. We started off with a drummer and Marvin. Marvin wanted to do some work. I was sitting here and Marvin and the drummer, Bugsy Wilcox, struck up a groove while Marvin was playing an RMI keyboard. So I recorded it as a rehearsal on tape. That's how GOT TO GIVE IT UP was born.

HC: And then the cowbell . . .
AS: This is one Marvin's trade secrets. It was a juice bottle half-filled with juice. Marvin is hitting it with a spoon. (Laughter all around.)

Marvin is phenomenal. There's part of the score for TROUBLE MAN where he's hitting the headphones. He's a very creative cat — he'll play anything. Even on GOT TO GIVE IT UP, he's "playing" the console at the end of Part 2.

In working with guys like the Norman Whitfields, the Hal Davis', they all have their unique way of doing things. But Marvin and Stevie are the most unique. "If it sounds good, let's use it."

HC: I see you go for six mikes on the drum set.
AS: Usually I try not to use more than six mikes on drums. Even on a big kit with all the toms and cymbals.

HC: Because of the phasing?
AS: Right and because drums sound better natural. Once I was recording

Ollie Brown with his large kit and he felt I couldn't get a good sound with my six-mike technique. I asked him to bear with me and he became amazed at the sound I was able to get.

My favorites are C412s or 414s for overheads or snare, 545s on top toms, an 86 on the high-hat. A lot of it depends on what the drummer will play during the course of the song. He may not make the run of the whole kit. So I ride with the tune to see what he's doing. There's no point in having a lot of mikes open, sucking up noise if they're not being used.

HC: Did you specify anything special when you cut the lacquers for GIVE IT UP?
AS: Yes. It was flat at Motown. Jack Andrews, who cut the disc, is a capable disc-cutter. All he had to do was adjust to my levels and cut.

When I turn a tape in, the levels are all adjusted. That was part of my training from Detroit. The cutter doesn't have to constantly play around with levels and EQ. I really don't believe a lot in EQ, because after you've EQed yourself to death in the studio, then you take it up to the cutting room and EQ again just for the sake of doing it, it's ridiculous!

If you talk to the average independent producer, the first thing they talk about after they mix is how much high and low they're going to add in disc-cutting.

HC: Do you ask them why they don't do it when they mix?
AS: Everyone's locked in to EQing the tape. "When you cut, you *automatically* add some bottom and top." I think that's ridiculous! To do it just for the sake of doing it — it's just a crutch, like dope, you've got to have it. My thing is to turn it in and cut it flat. That's it.

HC: Is the album mix the same as the single mix?
AS: Exactly. I made a tape copy from the master 2-track for the 45.

HC: First of all, I haven't heard the LP version, but I have the single and there seems to be a little extra high-end there.
AS: It could be because of the difference between cutting a 45 and cutting an album — the mechanics — and that will give you the *impression* that it is different.

HC: I guess it's Fletcher-Munson at work.
AS: That's what it is. I didn't use any EQ on I WANT YOU either. Like I say, I don't like to use EQ on vocals — maybe on instruments and background vocals. On my own, I've never EQed a voice.

HC: What about when you mix?
AS: In a mix, that's something different. I've never seen anyone yet EQ a vocal, then not *re-EQ* when they mix. If I sit down with a producer, record drums, and EQ

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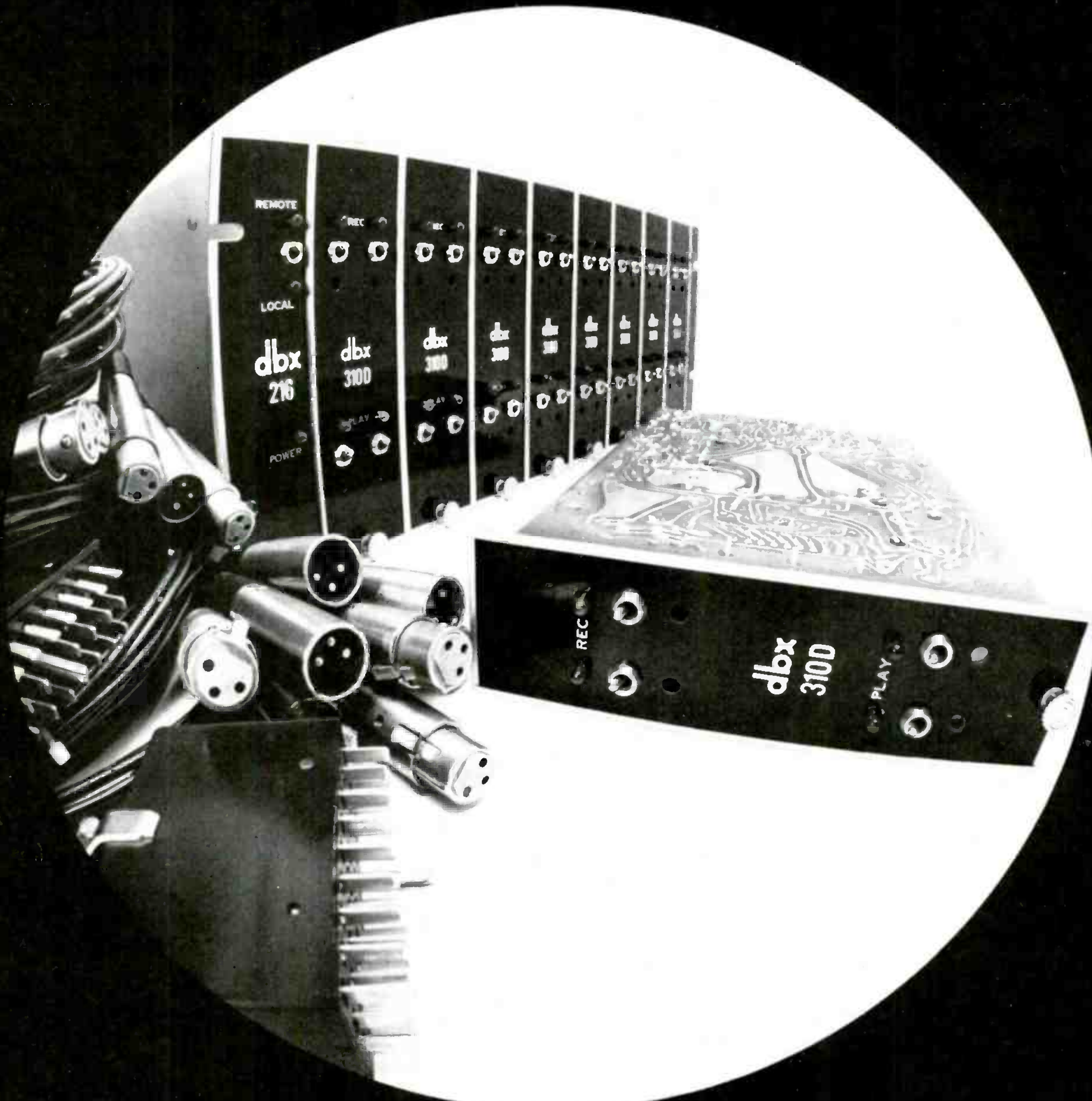

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when I *record* to the producer's satisfaction, those drums are going to be *re-EQed* after the strings, horns, and vocals are recorded. So you get up into *double* EQ. Why get into all that noise? It gets to be a drag. You drag it out to where you're really in trouble.

We took a version of GOT TO GIVE IT UP to a disco to play it. They really loved it there.

HC: Do you have a special technique you use for a disco mix?

AS: You have to treat it a little differently. Someone, and I wish I knew who, came up with a "design" for a disco mix. A friend of mine took a record of a singer I'm co-producing, Rick James, YOU AND I, to a disco. People weren't familiar with it, but by the time it got to the groove section, everyone was dancing. There was only one bad comment. "The disco mix is not really a disco mix because you didn't let the record break down into bass drum and congas." So I said to myself, "Self (laughs), who in the world would care if that was happening?"

HC: You know the Bee Gees song YOU SHOULD BE DANCING? There's bass drum and congas in the solos and that's considered to be a good disco song. Same thing with The Hollies' DRAGGING MY HEELS when they re-mixed the record for disco release. They featured the tom-tom rolls and keyboard work.

AS: I re-mixed GOING DOWN TO LOVE TOWN by the Originals. They loved that, the way I brought the congas up in the 45 release. But I didn't know anything about the album release — I brought the congas up inadvertently because that's the way I *interpreted* it for a single mix. I didn't know the formula, but now I do.

If we have a disco song on Marvin's album, I'll drop everything out, leave the bass drum up for a minute, give them some congas, then get back into it and they'll love it! I'm even finding out they count beats per minute!

HC: Does this go back to Dick Clark, "I'll give it an 85."

AS: You have to remember that formula; the bass drum, the congas, and the strings.

HC: "Dancing strings" I call them.

AS: Exactly, that's it. You've got those three ingredients and you're in.

HC: Do you like to use a lot of limiting and compression?

AS: I'm not opposed to limiting. The only time I have used compression is when I wanted to alter the sound. I would throw a limiter across a vocal. I can do Marvin without a limiter. But I would set a limiter up and have it in the chain, whether I use it or not.

HC: Just set the threshold to where it wouldn't trigger unless it was a horrendous passage?

AS: Right. Now to use a compressor would alter the dynamic range, which I don't like to do. It changes the sound between the top and the bottom.

You can also limit the output of the console, and there are many pros and cons on that. But if recorded

STUDIO SET-UPS

DATE 3-20-78

PRODUCER Various

SUNSET ☐ SUNRISE ☐

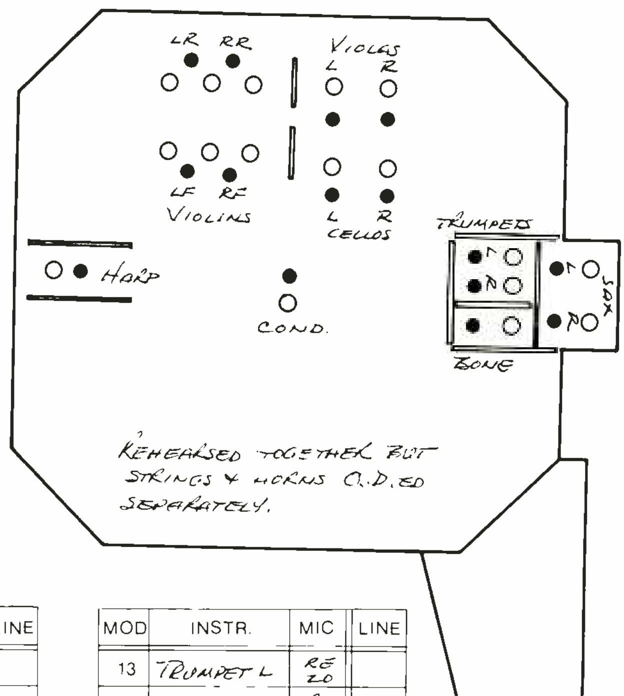
TIME 9:00AM Start

ENGINEER Art Stewart

ASSISTANT Tina Madison

COMMENTS Good Luck

TV Themes
· Sweetening ·
"James at 16"
2 Trumpet "Carter Country"
2 Sax "8 is Enough"
1 Bone "Maude"
"Good Times"



MOD	INSTR	MIC	LINE
1	Violin LF	86	
2	Violin RF	86	
3	Violin LR	86	
4	Violin RR	86	
5			
6	Cello L	87	
7	Cello R	87	
8			
9	Viola L	545	
10	Viola R	545	
11			
12	Harp	86	

MOD	INSTR.	MIC	LINE
13	Trumpet L	RE 20	
14	Trumpet R	RE 20	
15			
16	Bone	SM 56	
17			
18	Sax L top	86	
19	Sax L Bottom	86	
20	Sax R top	86	
21	Sax R Bottom	86	
22			
23			
24	Comp	87	

CUE A Strings

CUE B Horns

CUE C Cond.

properly, you shouldn't have to do that. I automatically run a buss through a pair of limiters to where they do nothing until they reach a certain threshold. If I'm doing a mix and a horn line comes up that would be a problem, the limiter should be able to catch it. Sometimes guitars are really problems. If you have more than one, which you usually do, then I would use some *peak* limiting. Once an LA-3A starts *compressing*, I would immediately re-set it and start the mix over again. I would never let it go into compression. To me a "tight mix" is too sterile. It's *too* tight. The music should flow.

HC: Well, I notice on the single of GOT TO GIVE IT UP, the needles bounce.
AS: Yeah, you've got to have that.

HC: It "breathes".
AS: That is the key. If you listen to any of the mixes I've done, you'll find that the tracks are like that. Unless you're working with a producer who is very adamant about what he wants — they're the worst type. Because after you've gotten a mix and some line comes along that he really wants to hear, he'll go haywire. It'll just destroy everything because he wants to bring that lick up. To me, that isn't right. You can do it, but you have to do it with reservations. You can't use a lot of fader travel on the board, just use a little bit and it will be noticable. You'll be able to hear enough of the lick.

When a producer listens to a song, he is listening to one certain *part* of the song, while I'm listening to the *overall* picture. With multiple 24 tracking, you cannot put everything a producer or arranger wants into the song at *one* level at *one* time. *It can not happen.*

HC: Everything can't be a solo.
AS: You do it to the best representation of what you can get. If you have to sacrifice something, you do it.

HC: And noise reduction?
AS: Without saying who technically has the best system, I would say dbx has the best system, based on my ears. But Motown is locked into Dolby. dbx seems to give back more

of what you put in. So if I *had* to use something, I would probably use that.

At the conclusion of Marvin's album, I'm going to outfit the 2-track machines with dbx cards and do one dbx mix and one non-Dolby simultaneously. Bill Ravenscraft, the tech supervisor of the studio, and I will then evaluate them. He feels that dbx is better than Dolby and dbx is far better than non-Dolby.

Then he and I can lay that to rest because I'm very much a non-Dolby man. If I find that dbx has not altered the sound that much and the mix still sounds "alive", then I would do the album dbx. I'm definitely closed on Dolby but a bit open on dbx.

HC: Would you rather go 15 ips non-Dolby?
AS: Yes, for the type of music I record. Speaking for the acts I produce myself, they're all 15 ips non-Dolby. I'd rather have a little tape noise with true fidelity and dynamic range than a tape without any hiss, which sounds stifled.

HC: How about the 30 ips debate?
AS: 3M's going to hate me . . . 30 ips is a waste of tape.

HC: A lot of people like the high-end and the signal-to-noise.
AS: What about your low-end? I could put up with 30 ips because the hiss is decreased somewhat but the bottom sounds a little flat. At that point it's a technicality because the consumer could give two hoots in hell if you did it at 20 ips, 90 ips, or 10½. At that point, we're satisfying ourselves because we're involved in that environment. But that fight will be around forever up to and including the digital process.

We did a test at Motown a year or two ago. They brought in digital vs. tape, and tape beat the digital hands down. The digital system was so clean, it was ridiculous!

HC: It sounded like computer-robot?
AS: It sounded like robot-mechanical music.

HC: How about these vocals you're recording in the control room?
AS: It takes a lot of patience and a lot of talent on the part of the artist. You're working strictly with headphones and it's very critical. We started the process on the I WANT YOU album and we're continuing it on this LP also. Recording in here is a headache though. You always have someone popping through the door because with Marvin, you never had a closed session — the doors are never locked because he feels better with people around. But that makes it difficult for me because you can hear the punch-ins and the machines running on the Sennheiser 415 that we're using.

He's also like Levi Stubbs in that he can match his voice very well. The average person can't relate to level through headphones, but Marvin can punch-in a spot and if I have him set on a fader, he stays there. He will tell you if it sounds too loud!

HC: Now he's recording through the cans, that means you have to do the same thing. Isn't that a pain for you?
AS: No. They don't bother me. I can even hear better with the cans because I can listen at low levels. The way some people monitor these days, the levels can drive you crazy. Even when we do overdubs, we do them at relatively low levels.

HC: Would you ever mix on cans?
AS: Hmmm . . . that's something I'd have to get used to. I believe I could though. Sometimes in cans, the sounds are more defined and clearer whereas in the monitors the sound might get lost.

Unfortunately, there's not enough attention given to mike technique. There's not enough guidance from the producer or engineer to the artist in the proper use of a microphone; as in the areas of diction, breath control, too much air in certain words, and the distance from the mike. These should be easy to control. You should be able to say "When" without saying "Wwwhen".

The same thing is true with loud headphones. They go hand in hand.

HC: You mean for cues?
AS: The level in the cue systems. Usually a guy will start out wanting them too loud and that's the most detrimental thing to your ears. It's a hindrance and pretty soon, you can't hear.

Mike technique really would enhance the vocal if you knew when to turn slightly, when to move in, when to move back. My advice to the new artist is: Treat the microphone as if it were your lover's ear. You wouldn't yell in it. If you wanted to whisper, you'd move in closer. That's the way to look at it.

HC: How do you like these Westlake-JBL-Universal speakers you're using?
AS: I love them, especially with the way this room is designed, you can hear them very well. You start bringing the tracks up, you don't get that masking effect.

HC: What about the 4320s you have here?
AS: They're just taking up space. I'm not knocking them, they're not to my liking.

HC: And the recording tape you're using . . .
AS: I'm not really into recording tape, but at Motown we ran a test of 250 vs. 206 ¼" and found the high-end was a bit squashed, maybe because of the biasing. So we stayed with 206 ¼", but use 250 2".

HC: How do you feel about consoles?
AS: Motown has a Quad-Eight console. I've worked on APIs, MCIs, Spectra-Sonics. To me, the API is a clean, compact console. I feel if a console worked to 80% of what a manufacturer says, you can work with it and make records.

We had some problems with this board and what they finally had to do was to put the modules in an ultrasonic vibrating bath until they got it straight. But consoles can become a matter of "What else can you fit in the board? What else do you want?" Automatic echo?

HC: Potato peelers?
AS: Right. My requirement for a console is that it be quiet.

HC: How about studios?
AS: Studios are like consoles — they're no big deal. Needless to say, all rooms are different. You have to go in and listen, hopefully take a tape in, do a little recording or mixing, take it home, listen and judge. But if it's a one shot deal, you have to rely on the people that are there to tell you how it will be. If you ask them if the speakers are flat, and they tell you yes, you know the speakers are not flat. But proprietors will tell you they are. *I know* the speakers we have here are flat to the specifications of the manufacturers.

HC: Which might be 12 k and down 3 at 15 k?
AS: Whatever they are, that's what they operate at. When Motown first installed the speakers in here, they asked, "How do you like them?" I said great. But then they said, "We haven't tuned the room yet or equalized the speakers." I told them, "Don't." And they haven't. They're sitting here just like they installed them.

HC: So there's no compensation for the dips and peaks?
AS: They brought the graphs in and the whole trip. One of the chief engineers at Motown told us, "The top-end is about +3 above where it should be. We want to bring it down." So I played with that for a couple of days, didn't like it, and had them come back and do it the way it was before. I've gotten more compliments for this room from my peers. I've gotten more compliments for I WANT YOU. They say the album sounds like an album should sound — the clarity's there. You can hear what's happening.

HC: What work did you do on I WANT YOU?
AS: I did the mix and some of the original recording. The work was started at Motown while Marvin's studio was being built. By the time it was finished, the album was ready to be mixed and we brought it to Marvin's. The tracks were laid down very well, therefore it was easy to mix. Because of the design of the mix room, everything was clearly defined.

HC: What constitutes "easy" tracks for you?
AS: Easy tracks are not overmodulated or distorted. They seem to "balance" within themselves and should be recorded at a level respective to what people are playing. For example, a tambourine has a lot of high-frequency information, so you wouldn't record it at 0 VU. But if you were to record a guitar at zero and then place them at zero on your monitor system, the *relationship* should be maintained in *level* perspective. To me, that's good recording.

HC: Could you go into your piano techniques?
AS: At Motown they have a few pianos. One is a seven foot and one is a ten foot Grand. I have various techniques depending on who's playing and which piano is being played. If you want to get into a stereo situation, you can use a top and bottom mike. I don't like using two mikes all the time because I don't feel it's necessary all the time. Sometimes stereo piano is an absolute waste. But if a person has a very light touch, you can use a KM86 over the top portion of the sounding board and the bottom. If you're playing with a heavy-handed player with a lot of dynamics, you have to use dynamic mikes to handle the level. A 57 or 635As would be good. For a lot of top-end, I like a 451 for its bite and an 87 on the bottom for that extreme definition between top and bottom.

On the large piano, I've found it to

be advantageous to take two 635s or RE15s and cross them at the neck and mike them right in front of the hammers. If I have to use three mikes for the large piano, I'll use an RE15 at the top, an 86 in the center, and an 87 at the bottom. If you have phasing problems, you have to do some shifting.

HC: Do you like to use pads over the piano?

AS: It definitely ruins the sound. You have to try to compensate with some EQ to try to restore the resonance. I would prefer to make that slight sacrifice than to have a piano full of leakage.

HC: How come Motown isn't into more remotes?

AS: I don't know. They have live tapes on everyone.

HC: They had Stevie Wonder's FINGERTIPS back in '63 and they probably had stuff from the Apollo.

AS: I think they should do more. As a matter of fact, that's an area that I'd like to get into. I've gone to some concerts to see people like Stevie Wonder and Marvin Gaye and the sound was just kind of blah. Yet I've heard some systems sound fantastic. I'd like to get into it after some OJT. Maybe it's because some of the guys that do concert sound get caught up in the concert and he's not operating the console. The console just sits there and does its own thing. He just sets it up for a rehearsal and that's it. *You have got to be attentive* to the sound at a concert. What if the artist should decide to deviate from the song. He may decide to jump over to a background mike and do a verse from there. You'd have to be able to raise the level of that mike and follow it. I've never seen this done. I'd like to give it a try and have Marvin Gaye and Stevie Wonder sound like they should. The artist should also be cognizant of better sound. I can understand a mixer getting caught in the passion of the moment, but to let it go bad the whole night long — there's no excuse.

HC: Since you're a producer also, do you find it hard to engineer while you produce?

AS: It makes it very difficult. In the projects I have done on my own where I have to be everything, I try to get someone to help. As an engineer, you have to be looking for distortion, noise, levels, etc. But as a producer you're looking for bad notes, performances, etc.

Personally, I couldn't care less who engineers. I don't need a "name" engineer, just someone who's proficient at his job. I don't want to get caught up in that superstition of the business where you have to have certain engineers, or musicians, or studios all the time. I don't want to get caught up in the know-it-alls. Each day brings about something different and new. You can't go on cutting the same way all the time.

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STABILIZING OPERATIONAL AMPLIFIERS

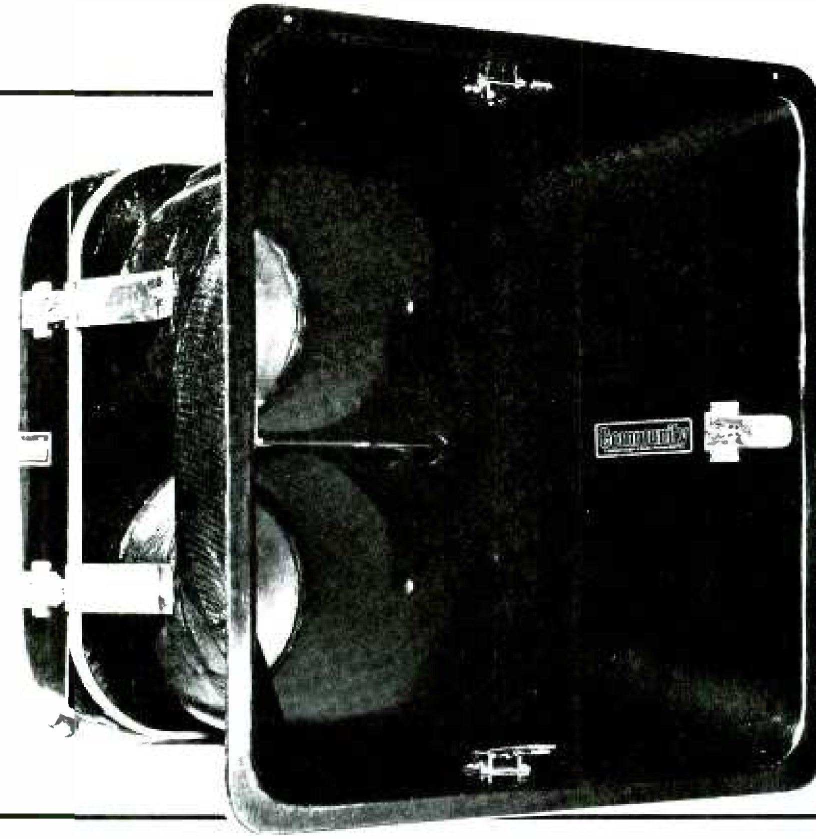
by
Deane Jensen

Merits of Stability

For the scope of this text, stable operation of a feedback amplifier decribes a condition which exhibits no oscillations, no overshoot, and no peak in the closed loop response. Of course, equalizer and filter circuits can have overshoot and peaks without instability; these will be discussed after the general case of the wideband "flat" gain block. Stable operation also results in relative freedom from RF sensitivity and the type of clicks and pops which are actually bursts of oscillation triggered by an external transient noise source. Good stability and low overshoot will result only if the feedback circuit response determines a reasonable amount of feedback at all frequencies of open loop gain. Often that means up to 10 MHz. Low overshoot and ringing means freedom from transient distortion caused by oscillations trailing each steep slope of the waveform.

"Unity Gain Stability"

Amplifiers differ in their ability to be stable even if the external circuitry is optimum. To evaluate the stability potential for a particular amplifier type, graphic data is required for both the "gain vs. frequency" and "phase vs. frequency" open loop performance. If the phase response exhibits -180 degree shift at a frequency where the gain is above unity, the negative feedback will become positive feedback and the amplifier will actually sustain an oscillation. Even if the phase response is less than -180 degrees and there is no sustained oscillation, there will be overshoot and possibility of bursts of oscillation triggered from external noise sources, if the phase response is not "sufficiently less" than -180 degrees for all frequencies where the gain is above unity. This "sufficiently less" term is more properly called "phase margin". If the phase response is -135 degrees, then the phase margin is 45 degrees (the amount "less than -180 degrees"). Actually, the "phase margin" of interest to evaluate stability potential must also include the phase response of the feedback circuit. When this combined phase margin is 45 degrees or more, the amplifier is quite stable. The 45 degree number is a "rule of thumb" value and greater phase margin will yield even



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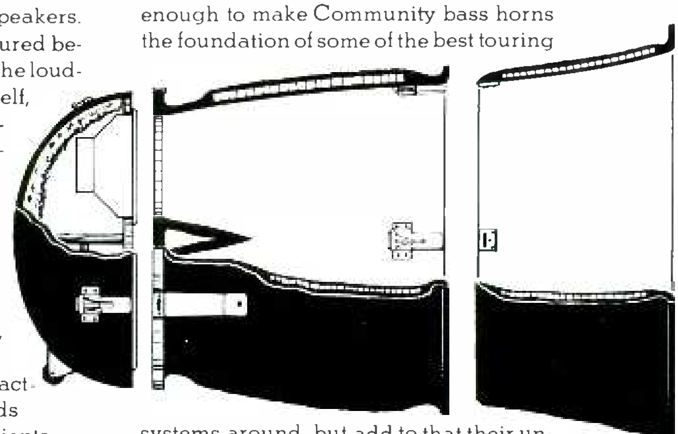

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SPECIFICATIONS	EXTENDED LEVIATHAN	BLT	FRC/B
Flare Rate	48 Hz	52 Hz	66 Hz
Operating Range	from 50 Hz	from 60 Hz	from 75 Hz
Driver	Two 15"	One 15"	One 15"
Size (HEIGHT/WIDTH/DEPTH)	43¼"/69¼"/64"	44"/44"/56"	30½"/40"/44"
Weight (less drivers)	175 LB	90 LB	65 LB

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better stability and less overshoot.

Often, but not always, the lowest phase margin is at the highest frequency which has gain above unity; because there is always some delay independent of frequency which represents more degrees at higher frequencies. An amplifier with 45 degrees phase margin at the frequency of unity gain open loop is said to be "unity gain stable". Some amplifier types can be compensated for unity gain stability optionally at some sacrifice in slew rate. If stability is considered to be of high priority, the tradeoff must be made. "Unity gain stability" means stable operation at the lowest closed loop gains where stability is usually worst.

Miller Compensation Zero

If the amplifier is the Miller compensated type with a capacitor from collector to base of the second stage, a resistor is sometimes used in series with the "Miller compensation capacitor" to create a zero in the response at the frequency of unity gain open loop. This adds up to 45 degrees to the phase margin at the frequency of unity gain open loop. A Miller compensated amplifier without the zero will most probably exhibit less than 45 degrees phase margin at the frequency of unity gain open loop and therefore it may not be "unity gain stable".

Excess Phase

There is always some delay in the amplifier which is independent of frequency and will therefore cause the amplifier to exhibit an excess phase component which appears as an increasing number of degrees with increasing frequency. This delay is critical term relating the frequency of unity gain open loop to phase margin. Usually the amplifier response is close to a single pole or 6 dB/octave which creates a phase response near unity gain of -90 degrees. If the frequency-independent delay represents -45 degrees equivalent phase response at the frequency of unity gain open loop, the total is -135 degrees or 45 degrees phase margin. The number of degrees of "excess phase" created by the frequency independent delay is dependent upon both the delay time and the frequency of unity gain open loop. Realizing that only 12.5 nanoseconds excess delay limits a 10 MHz amplifier with a single pole compensated rsponse to 45 degrees phase margin, explains why stability and unity-gain bandwidths higher than 10 MHz are usually mutually exclusive. Given the excess delay time (T), the maximum possible unity gain frequency (Ft) for 45 degrees phase margin can be calculated: FT = 1/8T.

Feedback Compensation

So far, the characteristics mentioned relate to the internal amplifier circuitry. And the analysis of the phase response reveals only its potential to realize a stable gain block. For the scope of this text, only amplifiers which are unity gain stable will be considered acceptable for general applications. Amplifiers, which are not unity gain stable, require analysis beyond this text and are usually confined to fixed gain configurations; as the required compensation must be changed for various closed loop gains.

Recall that the phase margin of interest for analyzing stability includes the effect of the feedback circuit phase response of related delay. Another important viewpoint which suggests the need for frequency compensation in the feedback circuit is revealed by the "Bode Plot"(refer to Figure 1) of the open and closed loop responses. The graph shown is a simplified plot showing the open loop gain of a typical operational amplifier with a gain bandwidth of 10 MHz (some amplifiers have a lower GBW product which means the plot must be moved to the left). Amplifiers differ also in the exact shape of the open loop response. Also for simplification, the "closed loop" plots shown here are actually the inverted feedback function which do not include the effect of the limited open loop gain. Generally the open loop gain diminishes similarly to the 6 dB/octave slope as shown, which describes some real limitations to the amount of gain which can be realized as a function of frequency. Note that a 10 MHz amplifier can exhibit 40 dB of closed loop gain only up 100 kHz.

This point is called the "frequency of intercept", where the uncompensated inverted feedback function (1/B) intersects the open gain plot. Since the amount of feedback at any frequency is approximately the difference between these plots, there would be no feedback for the frequency range above intercept if a simple resistive voltage divider is used in the feedback circuit. A capacitor Cf connected across the series feedback resistor Rf can be used to ensure a finite amount of feedback for the entire frequency range of open loop gain greater than unity. The pole

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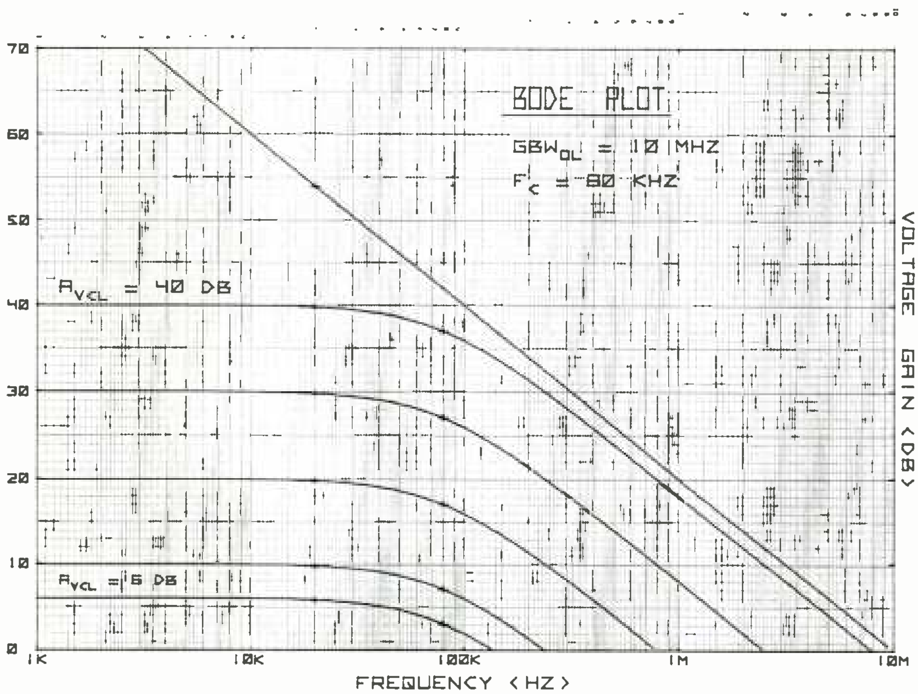

frequency created in the closed loop response must be lower than the frequency of intercept by the same ratio as the feedback voltage ratio chosen for the frequency range above intercept. If 3 dB is considered initially, the inverted feedback function pole must be at the frequency of intercept divided by the square root of 2. Higher values will improve stability and reduce overshoot with reduced bandwidth.

Then converting to the time constant form by:

Tc = 1/(2*PI*F)

The capacitor can be calculated by:

Cf = Tc/Rf

The calculation should be used to determine an initial value, but observation of the small signal overshoot should be used to finalize the compensation. This type of compensation is called "Feedback-zero compensation" or "Phase lead compensation" since it creates a zero in the feedback circuit which advances the phase of the feedback signal.

Variable Gain Control

The family of closed loop response curves (inverted feedback functions) in the "Bode Plot" shows a variable gain amplifier controlled by adjustment of the feedback shunt resistor Rshunt. The series feedback resistor Rf and the compensatory capacitor Cf are fixed. This method yields close to "constant bandwidth" over the range of gain adjustment. The calculation of the compensatory capacitor value must be made for the maximum closed loop gain. Usually, Rshunt is realized with a potentiometer in series with a fixed resistor which sets the limit for maximum closed loop gain. If the fixed resistor is connected directly to the inverting input of the amplifier, it will isolate any capacitance associated with the wiring to the potentiometer from causing a delay in the feedback signal which would reduce phase margin.

If the series feedback resistor Rf is adjusted to control gain, the bandwidth increases as the gain is reduced. Generally, this method may exhibit more overshoot at low gain, and would be used only if the bandwidth determined by the above calculations is not considered sufficient at lower gain. Since the gain-bandwidth of this method is maximum at low closed loop gains, the transient response must be observed at the lowest gain setting to determine the maximum allowable gain-bandwidth which can be realized safely even though the initial calculation is made for the maximum closed loop gain.

Unity Closed Loop Gain

Even with a stable amplifier, unity closed loop gain operation results in the highest overshoot, sensitivity to RF pick-up, and possibility of bursts of oscillation triggered from external noise sources. For stability, it would seem well worth while to arrange some gain the amplifier and add a voltage divider (pad) in the circuit to re-establish the overall gain of 1. Of course, this may increase the noise to an unacceptable level. Short of limiting the minimum closed loop gain to something on the order of 2 (6 dB) with some phase lead compensation, the merits of using a feedback circuit with a resistor paralleled by a capacitor (rather than a direct connection) should be evaluated in the laboratory for the specific amplifier type.

Input Low Pass Filter

A low pass filter at the input to the amplifier can be used to limit the bandwidth to avoid overshoot and desensitize the effects of stray positive feedback paths. This is done in addition to all other procedures, not in lieu of any necessary stabilizing precaution. It only affects the resulting response of the signal at the input of the amplifier. It does not affect all sensitivity to RF or external noise sources, so transient analysis, usually a good indicator for evaluating stability, may be optimistically misleading.

Source Impedance Effects

Of course, with high source impedance, the possibility of some positive feedback via stray capacitances suggests serious consideration of the input low pass filter. But at low source impedances, including summing amplifiers where the non-inverting input is grouded, an internal problem may occur. The gain-bandwidth of the first stage of the operational amplifier is high dependent upon and inversely related to the source impedance. This means that a lower source impedance increases the gain-bandwidth of the first stage. Consider the first stage as an amplifier which has in its feedback path the delays of the other stages as well as the feedback circuit of the complete operational amplifier. As the source impedance decreases, the first stage gain-bandwidth increases, and its phase margin decreases. In some amplifiers, the phase margin of the first stage can reduce below zero and the first stage will oscillate. If this amplifier type must be used with low source impedance, a resistor will be required in series with the input to limit the net minimum source impedance. Of course, this may affect noise. This problem is prevalent in amplifiers lacking emitter resistors in the differential input pair. The emitter resistors limit the gain bandwidth similar to, but not exactly the same as, the series input resistor.

The preceeding paragraph referred to the impedance at the non-inverting input. The impedance of the feedback circuit has another limitation regarding stability. If the feedback network impedance is high, additional phase lag

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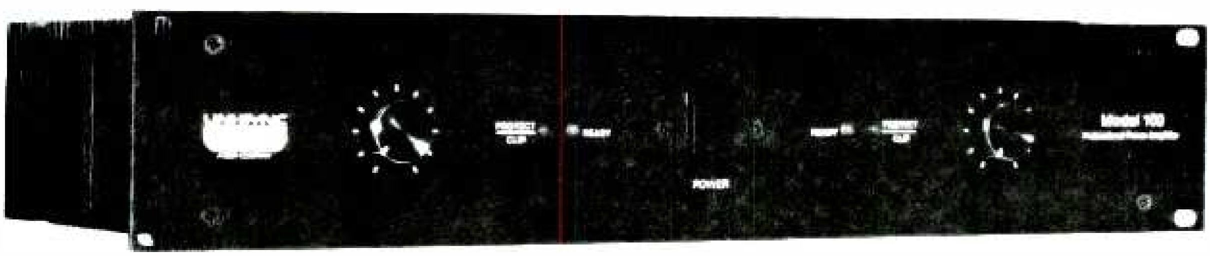

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or delay could occur as a result of the input capacitance. The resulting reduction in phase margin affects stability. This is more prevalent in amplifiers where the inverting (feedback) input is an emitter rather than a base of a transistor.

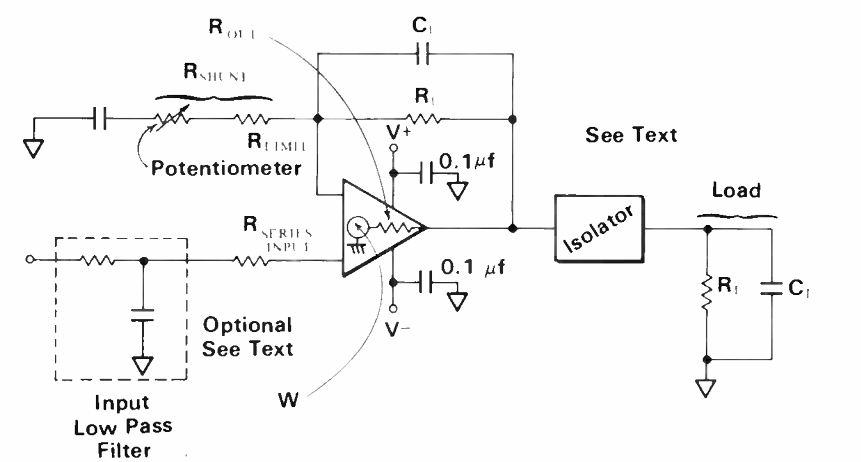
Load Isolation

The output impedance of an amplifier is not zero, but rather some finite amount which may actually increase with frequency. This condition is approximated by the schematic shown as Figure 2. This means that a capacitive load (even a length of cable) will cause a phase lag or delay of the signal at the output node of the amplifier. The feedback signal is derived from the same output node, so the feedback signal also suffers the delay caused by the capacitive load. As the capacitance is increased, the delay increases and eventually the phase margin will be reduced to the point of causing the amplifier to oscillate.

One remedy is a series resistor added between the output node and the load. The feedback must still be derived *directly* from the output node. The value of resistance required to isolate the effect of the load capacitance will depend upon the specific amplifier type, the load capacitance, and the closed loop gain. This value is best determined by observing the small signal transient response.

If the required value of resistance is too high compared to the output source impedance required by the application, an inductive series element can be used. An inductance exhibits low impedance at low frequencies where the low source impedance is usually required for the application. But the inductance yields increasing impedance with frequency which further isolates the capacitive load. An excellent load isolator can be made with 40 turns of #30 magnet wire wound

Figure 2: **Simplified Schematic Showing Components for Stability**



around a 39 ohm 1 watt Allen-Bradley resistor. The inductance impedance pole is about 40 uH creating a circuit zero at 155 kHz. So above 155 kHz the isolator is like a 39 ohm resistor, but below 155 kHz, the impedance decreases to about 0.2 ohms at DC.

A smaller isolator can be made by threading a piece of buss wire through a "ferrite" bead. This method can only be used with limited current levels, because the "ferrite" magnetic material will saturate and cause distortion at some maximum current. Of course, the current is proportional to level (voltage) and inversely proportional to the load impedance. Many different "ferrite" compounds exist including some Nickel compounds which exhibit low distortion over a wide range of levels. When the "ferrite" bead is used, an additional series resistor as small as 10 ohms very significantly improves the isolator. However, if the load is a Steel core output transformer, the 10 ohm resistor should not be used as the source impedance will affect distortion.

Some amplifiers have an internal series current limiting resistor in the output circuit in addition to the emitter resistors. This means the output impedance is high and therefore a capacitive load will cause more delay than with an amplifier without the additional internal resistor. This type of amplifier may not be suitable to drive a capacitive load, perhaps not even a length of cable without a relatively high value of isolation resistance. Since the feedback circuit also loads the output stage, the capacitor Cf required for "Phase lead compensation" may create a phase lag at the output node.

Power Supply Decoupling

Each amplifier must have a pair of low loss (low inductance) capacitors connected from each power supply terminal to the common point (ground) which is the reference for the load and the non-inverting input. This ensures a low impedance at high frequencies between these three circuit nodes. One small type of monolithic 0.1 uF capacitor is Centralab (USCC) #CY20C104P.

Active Equalizers

The inverted feedback function of an active equalizer must be analyzed to ensure sufficient loop gain at all frequencies. It is possible that the equalization function may describe a condition where the inverted feedback function exceeds the open loop gain at high frequencies. Computer modeling

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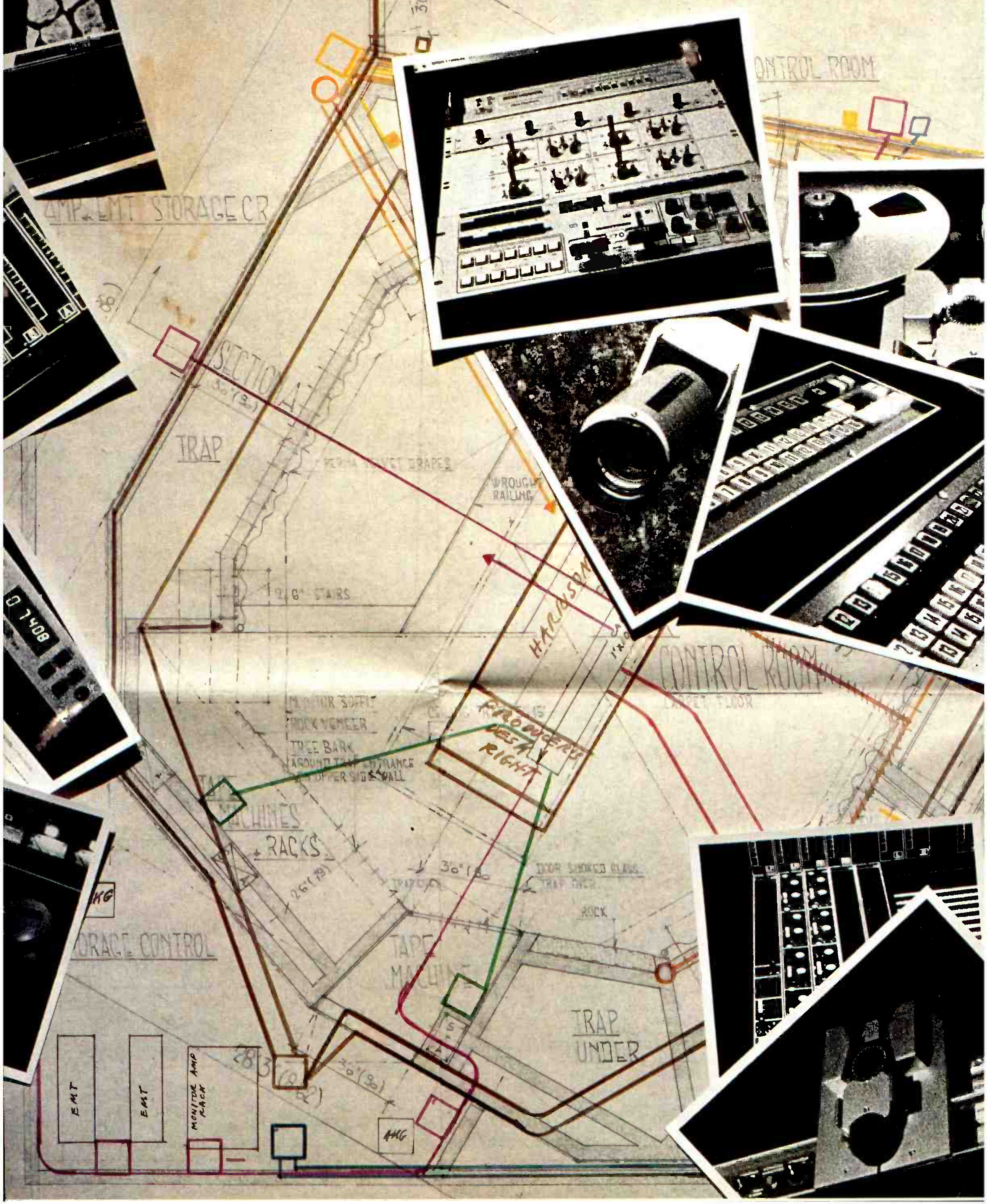

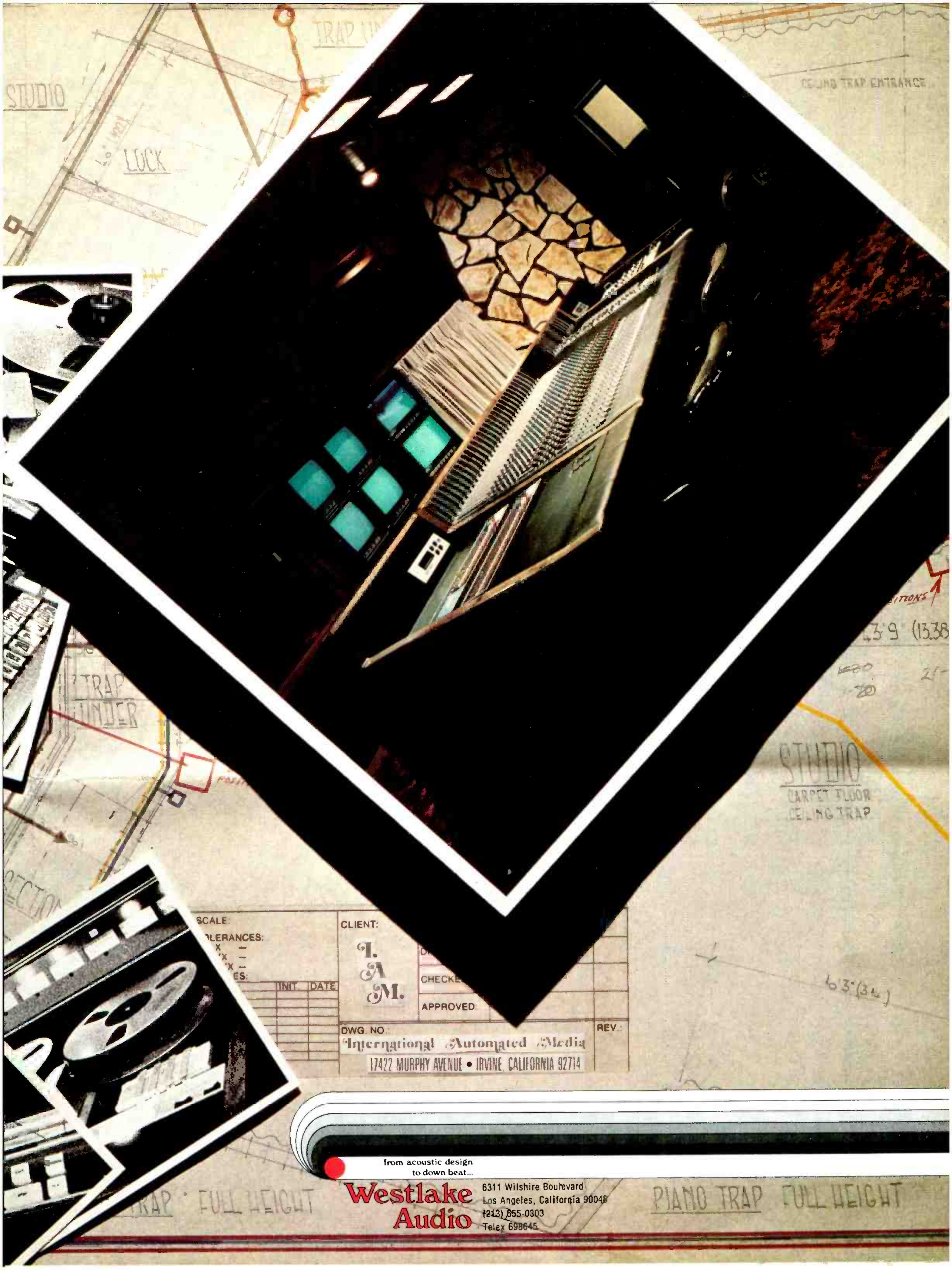
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is suggested to verify loop gain and phase margin as a function of frequency. Equalizers with switchable functions must be analyzed to reveal these functions for each switch position and "in-between-positions" to avoid clicks caused by momentary instabilities. Additional components to control stability for these "in-between-positions" can be considered after the response functions are revealed. The additional circuitry required can then be verified by computer modeling and observation of the small signal transient reponse.

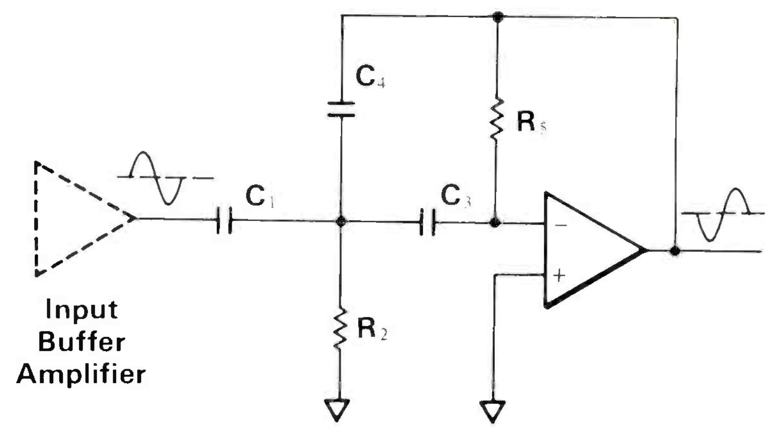


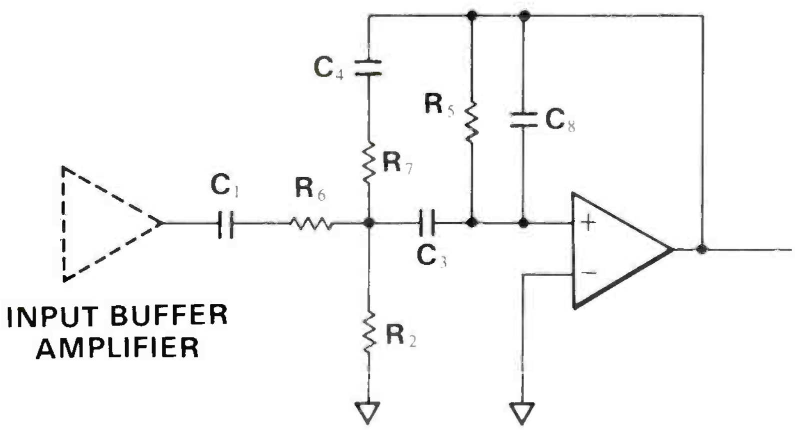
Figure 3: **Textbook Model For Inverting High Pass Filter**

Active Filters

Similar analysis is required for active filter circuits. A proper dose of skepticism applied to published topologies may improve stability. Computer-aided analysis of the active filter topologies published in popular texts has revealed that while the authors have accurately synthesized the described functions, they have not analyzed the resulting amplifier stability. Many of the popular topologies use unity closed loop gain configurations which are non-optimum for sensitivity to external noise sources. Many others, including the popular "inverting high pass filter" shown in the schematic, require individual analysis and modification for stability. Others have been proved to be "impossible to stabilize" without significant error in the function.

The schematic for the "textbook inverting high pass" of Figure 3 shows a topology which requires a voltage source (zero source impedance) at the input node to realize the function. Even a few ohms seriously alters the flatness of the very high frequency response. Therefore an input buffer amplifier is required and most all realizations in use

Figure 4: **High Pass Filter Model Modified for Stabilty**



$$R_6 = R_7 = R_2/10$$
$$C_8 = 0.1\mu s/R_7$$

today incorporate this required buffer. Note that the two amplifier outputs are anti-phase and are connected together through the two series capacitors C1 and C4. This condition results in peaking response due to the reduced phase margin caused by the phase lag created by the capacitive load. A second problem is a pseudo slew rate limitation (I/C) determined by the maximum output current capability of the amplifier and the capacitances.

The modified topology is shown in Figure 4 and requires two resistors R6 and R7 which isolate the effects of the capacitive loading and one capacitor C8 which is the phase lead compensation to avoid intercept. The value for the isolation resistors is chosen to desensitize the capacitive loading based on the peaking response and/or the slew rate limiting depending upon the amplifier characteristics. The two resistors R6 and R7 must be closely matched because their effects upon the very high frequency response are complimentary. The absolute value of R6 and R7 is set to one tenth the value of R2 to minimize the error which the stability modification introduces to the response function. A ratio greater than 10 will create less error. Noise and amplifier loading considerations should be used to scale the overall impedances used. The capacitor C8 avoids intercept by introducing phase lead compensation. The inverted feedback function of the "textbook" topology is 2 (6 dB) at all frequencies well above Fc of the function, so the intecept frequency was the gain-bandwidth divided by 2.

References:

Operational Amplifiers, Design and Applications, Burr-Brown; Graeme, Tobey, Huelsman: McGraw-Hill, 1971.

Operational Amplifiers, Theory and Practice, Roberge; Wiley, 1975.

SHIELDING, GROUNDING — AND SAFETY

by Ken Fause

Let us begin by demolishing a myth: techniques for shielding and grounding are not black magic; there are no incantations to chant while deciding where to connect Pin 1. None of the following information is by any means new; the techniques described here date to the early days of the telephone and cinema sound industries. Perhaps novel to the audio industry will be the suggestion for combination of audio system ground with AC mains power ground. Yes, Virginia, you *can* have a superior audio ground scheme which also complies with Electrical Code safety requirements, meets labor safety rules and makes your insurance company happy. While such an arrangement may be new to the audio field, it is frequently used in the research laboratory and for medical electronics.

In order to discuss the topic, it is necessary to go back to some basic principles. Here rests most of the confusion about shielding and grounding — the principles are so basic that they are often ignored. One disclaimer: we will be discussing general cases; no doubt many ingenious readers will conjure up situations where the proposed schemes will be inappropriate. What follows is *one* method which has been successful, there are others equally valid. For the interested (and determined) reader, several detailed references will be found at the end of the article.

Most professional audio activity takes place in modern, populated areas. Along with their many other attributes, populated areas are generally blanketed with stray electric, magnetic and electromagnetic (radio) fields. Some lucky few will record birdcalls in the wild. The rest of us are unfortunately faced with the problem of amplifying, processing, recording and otherwise manipulating audio program signals while rejecting hash, hum, crackles, CB radio chatter, television

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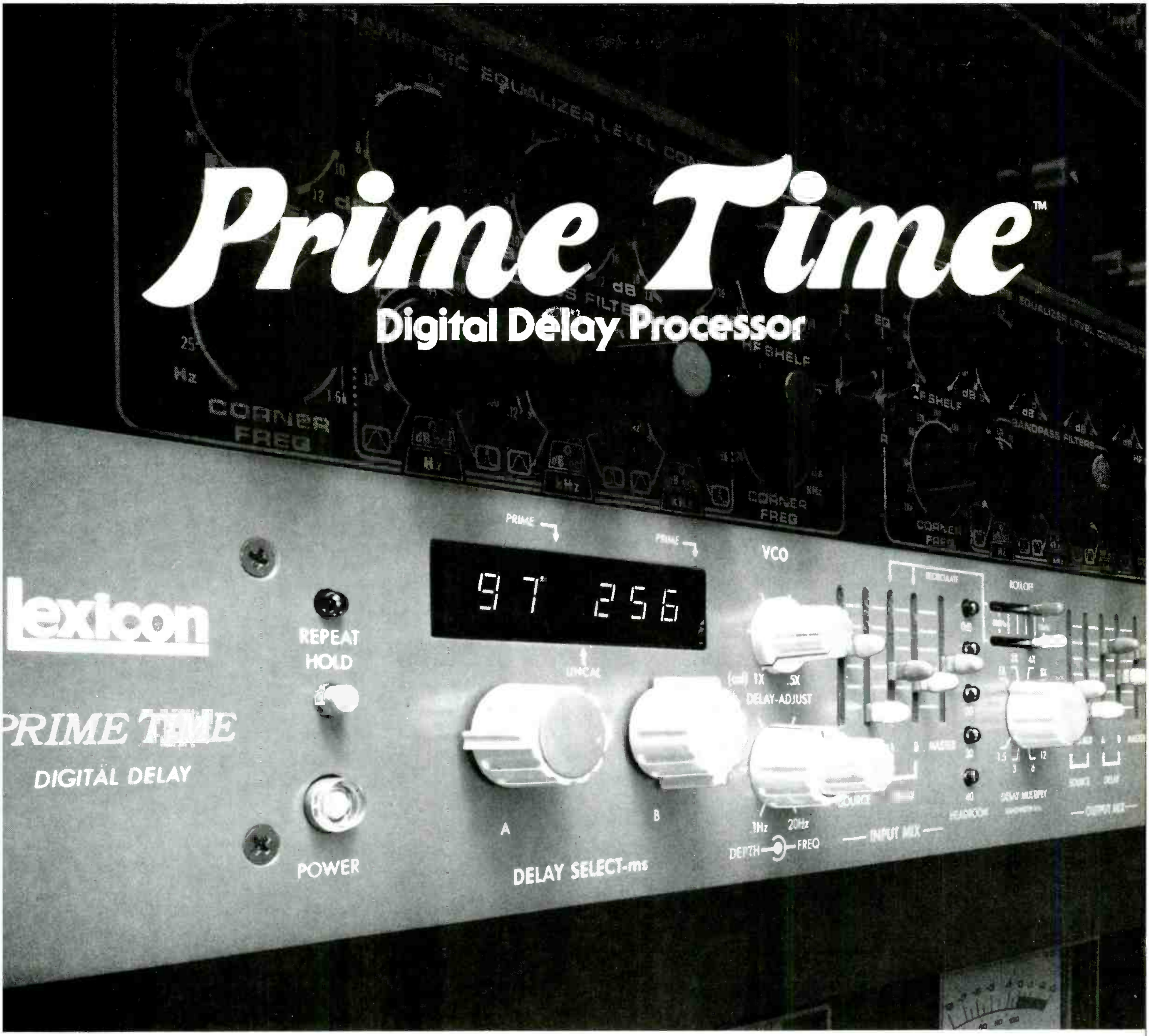

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sync buzz and other miscellaneous interference so readily contributed by the surrounding environment. For the sake of this discussion, we shall refer to all such interference as *noise*, where we define noise to be *undesired signal*.

In any situation involving noise suppression, be it acoustic, mechanical, or in this case, electrical and magnetic, we will be concerned with:

a) the source,
b) the propagation path,
c) the receiver.

In this case, the noise receiver would be the complete recording studio or sound system under consideration. In theory, one could achieve the desired noise reduction by attacking any of the three elements above. In practice, the choice is limited: most sources of interference are external to the studio, and usually one prefers not to modify expensive audio gear if at all possible. This usually leaves us only the propagation path to work with. In order to come up with an efficient approach, we must discuss some theory.

Types of Interference Coupling

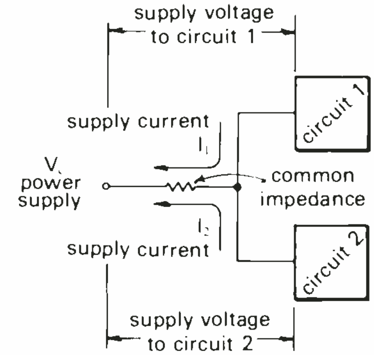
a) *Electric field* or *capacitive coupling;* the electric (charge) fields of the source and receiver circuits interact. (Sometimes called electrostatic coupling; this is misleading as the fields are usually time varying, not static.)

b) *Magnetic field* or *inductive coupling;* the magnetic fields of the source and receiver circuit interact.

c) *Electromagnetic wave* (Hertzian wave, radio wave) *coupling*. An electromagnetic wave is in fact a vector product of an electric field and magnetic field; therefore, when proper techniques are applied to suppress both electric and magnetic coupling, radio frequency interference (RFI) will most often be suppressed as well.

d) *Common impedance coupling.* Currents from two circuits flow through a common impedance. The situation may arise in power supply, signal and ground circuits, and is undesirable in all. An example is shown in Figure 1.

Figure 1: **Common Impedance Coupling**



The supply voltage to each circuit is affected by the supply current drawn by the other

It is essential to keep a basic fact in mind: *Whenever a charge moves in a conductor, that conductor will radiate both electric and magnetic fields.*

Electric Field Shielding

A *shield* is generally defined as a *metallic partition between two regions of space*. Infinite planes of shield in space are of interest only as textbook examples, in practice the shield is generally formed into an *enclosure* for circuit conductors or circuit assemblies. For special cases, an entire room might be shielded. The shield may be intended to *contain* the electric field (source shielding) or to prevent the ambient electric field from acting on some circuit (receiver shielding).

In receiver shielding, the essence of the idea is to create somewhere for the electric field charge to go *other* than the sensitive circuit or conductor.

For effective electric field shielding:

a) The shield material must be an electrical conductor.

b) The number and size of openings in the shield must be minimized.

c) The portion of circuitry or conductor extending beyond the shield must be minimized.

d) The accumulated charge on the shield must be drained. The shield should be tied to the lowest potential reference point of the associated electronics. In almost all audio uses this will be *ground*.

e) A shield should not carry other than its own charge drain current. Any current in the shield will produce corresponding radiated electric and magnetic fields; these will almost certainly couple into associated circuits as noise.

Grounding

Ground is commonly defined as a *zero signal reference point for a complex of electronics.* (Notice from this definition that a connection to *ground* and to *earth* are not necessarily the same thing.) Of course, reality is not so simple — a number of distinct functions all employ the term and concept of "ground". It is necessary to consider each.

Shield Ground. As stated earlier, this drains the accumulated shield charge. For the drain to operate, it must be

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connected to some large body which will sink the charge. In practice this "large body" usually is the earth. *Shield grounds thus must eventually connect to earth grounds.*

Transmission Ground. This provides a reference for signal voltages being transmitted. The most obvious case is a reference for single-ended (unbalanced) circuits. Balanced circuits usually do not *require* a transmission ground, although in common studio practice they are often *referenced,* but *not connected to* transmission ground. Transmission ground *should not* be the circuit shield unless the source and receiver equipment are especially designed for this usage. For example, Tascam equipment is so designed. Use of the special cable and connectors they furnish will optimize Tascam performance.

Circuit Power Ground. Exactly as the name implies. This is the zero potential reference for the circuit powering voltage(s).

Frame, Chassis or Rack Ground. Essentially performs the same function for a circuit enclosure that shield ground does for a conductor.

Safety Ground. Electrical Codes in force, labor laws, and the insurance carriers are virtually certain to require that all conductive enclosures and surfaces be bonded to earth ground by an approved means. The reasoning is simple. An electrical fault, such as an insulation breakdown between the hot side of the AC mains supply and the enclosure will make the enclosure live. An individual touching both the enclosure and ground will then be connected across the AC power line. This is not a hypothetical scenario — several performers have been killed by contact with "hot" microphones.

The solution is to bond the equipment enclosure to ground. There are two results:

1) A low impedance path to ground is established, much lower than that through the human body. In event of a fault, this creates a voltage divider, and a low body current will flow.

2) If a sufficiently low impedance ground path is established, then a very high current will flow under fault conditions. This will rapidly trip the circuit protection device (fuse or circuit breaker) and thus clear the fault.

Notice that the ampacity (current carrying capacity) of the equipment safety ground conductor is determined by code to withstand the maximum fault current that the AC supply mains can deliver.

Unipoint Ground Scheme

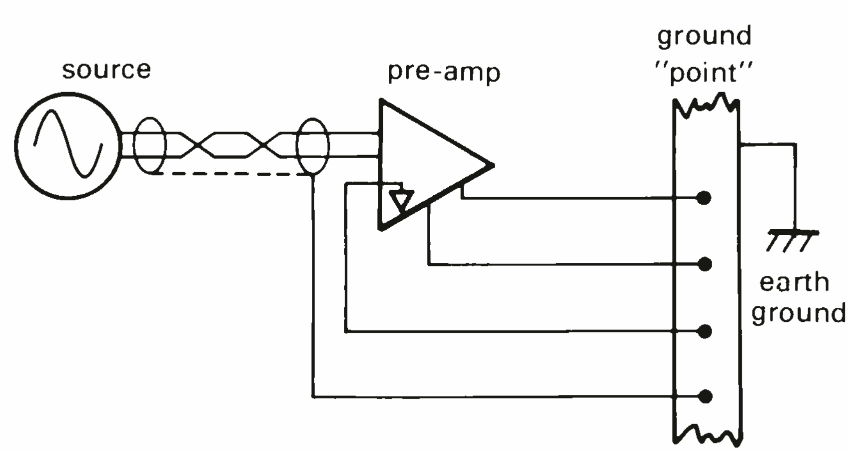
Each piece of audio equipment, and each type of ground associated with that equipment, connects to a single defined point by a single path. Separate bundles should be formed for ground leads of each functional type. See Figure 2 for a simple example. Strict application of this rule in large systems becomes a matter of rigor; any shortcuts taken will result in reduced system performance.

There are several very good reasons for this seemingly extreme approach.

a) At audio frequencies the concept of an extended ground plane is a polite fiction. Two separated ground points are seldom at the same potential, a potential difference will couple into the circuit as a noise voltage.

b) Separate ground leads for each ground type admittedly present a clumsy mechanical layout problem. The object is to avoid common impedance noise coupling as was

Figure 2: **UNIPOINT GROUND SCHEME**

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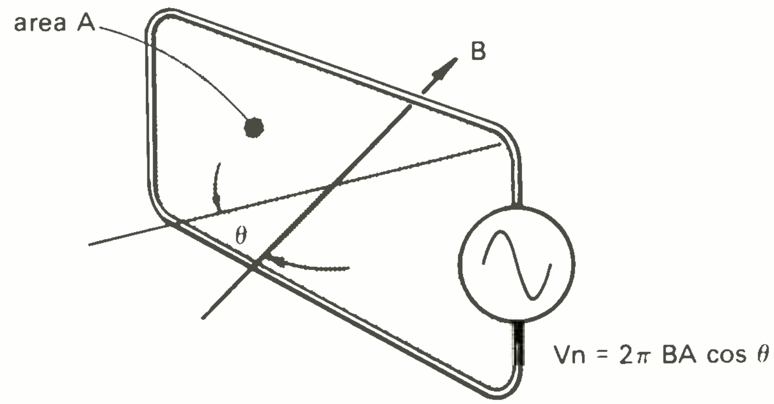
SEC
0 2 3 4 5
3 3/4
7 1/2

Figure 3: **MAGNETIC INDUCTIVE COUPLING**



described earlier. For this reason circuit power, logic power and lighting power should *not* share the same ground lead; switching transients are likely to wind up in the audio.

c) Separate ground type bundles may also seem extreme. Once again, recall that any current in a wire produces electric and magnetic fields, these in turn will induce noise in nearby circuits.

d) A time-varying magnetic flux will induce a voltage in a closed loop. This fact deserves extended discussion.

Magnetic Inductive Coupling

Please bear with the mathematics for a moment and consider the situation shown in Figure 3. Here we have a stationary loop of conductor enclosing an area A. Presume that we thread this loop with a time-varying magnetic flux of density B constant over the area. It may be shown that the induced noise voltage is:

$$V_n = 2\pi BA \cos\theta$$

where:

A is the area of the closed loop.

B is the rms value of a flux density varying sinusoidally at frequency f.

θ is the angle between the flux vector direction and the plane surface of area A.

Vn is the rms value of the induced noise voltage.

Hang in there! The object of the game is to minimize Vn, the induced noise voltage. Everything on the right of the equation is a product; thus, we may minimize Vn by minimizing any or all of the terms on the right. This is the basis for most methods of hum suppression.

Our flux field is likely caused by stray AC mains radiation, so we are not likely to change f. Both B and θ may be changed by altering circuit conductor locations, moving transformers, and the like. *The most powerful technique is to minimize the loop area A.* This is the main reason for tightly twisting the signal leads of a balanced transmission line.

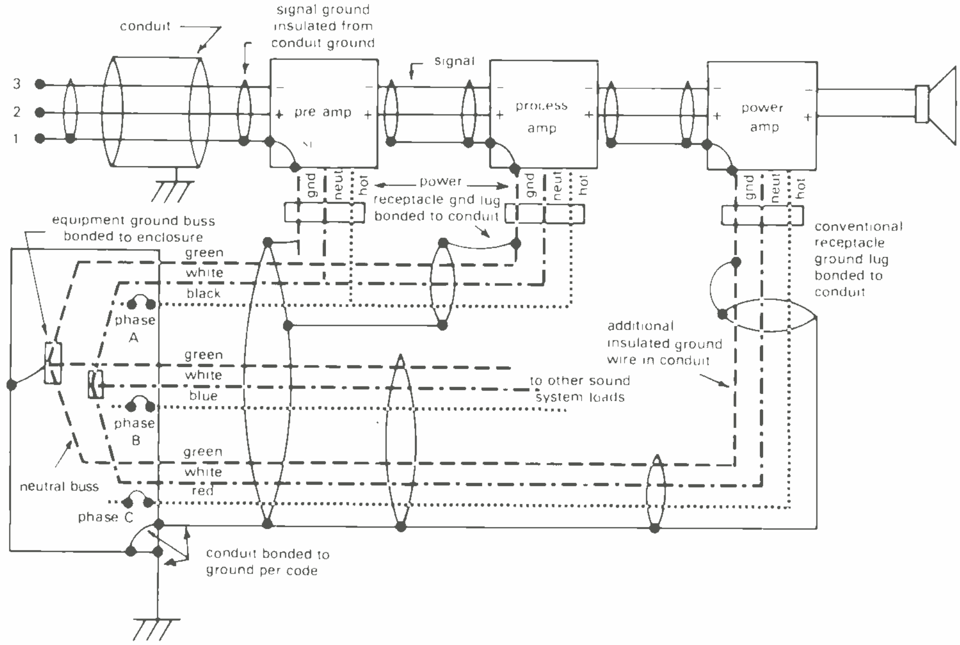
One last, critical point. *If the conductor loop is not closed, there is no enlcosed area A.* If A = 0, Vn must = 0. *No loop; no magnetically induced noise voltage. This is the primary reason for a unipoint ground scheme.*

Having very carefully made this point, we now proceed to state the exception to the rule. The unipoint ground scheme, for all the stated reasons, is strongly preferred for use at frequencies up to about 1 mHz. Where cable lengths approach 1/20 to 1/10 wavelength and where high radio frequency (RF) fields are present, it is often necessary to ground the shield at more than one point to insure that the entire shield remains at ground potential at all frequencies. A capacitor is used to establish the required RF ground; disc ceramics of .01 mf have proven effective in practice.

A final point about magentic induced coupling: *A grounded shield around a conductor does not attenuate magnetic induced noise at audio frequencies.* The skeptic is invited to try a simple experiment. Set up a properly shielded and grounded circuit as described. Place a bulk tape eraser or transformer-type soldering gun nearby and turn on the magnetic field. Measure or listen — but if you listen, use extreme caution with monitor levels, this is an excellent way to destroy woofers. Only magnetic materials such as steel or mu-metal will provide a useful degree of magnetic shielding in the audio frequency region.

Grounding and Safety

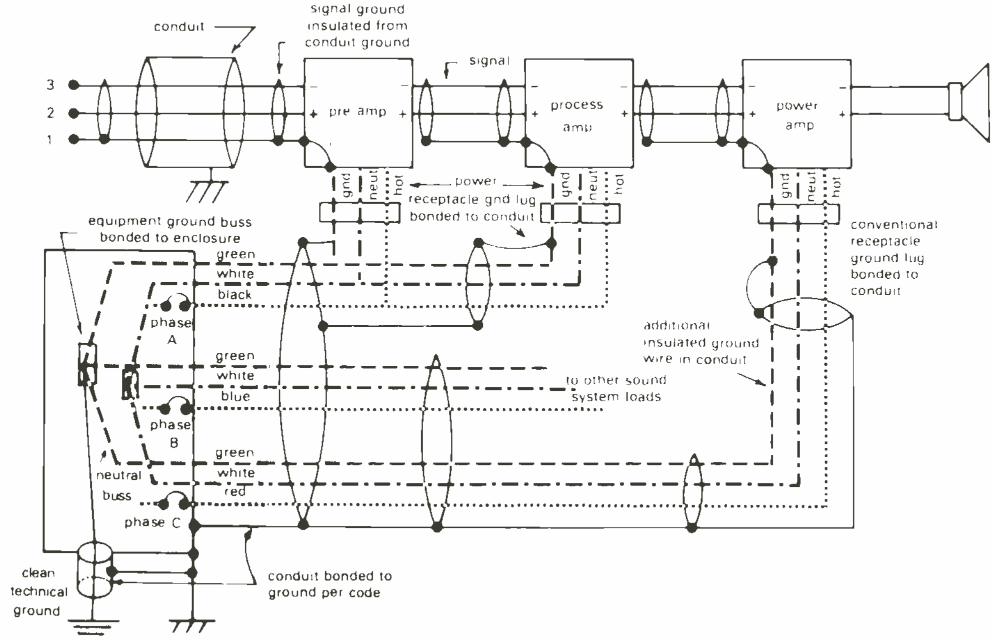
A potential conflict of safety and function exists when sound equipment must be installed which is manufactured as is common with chassis ground, audio signal ground, and equipment safety ground tied together. One solution to this quandary has been the abhorrent, but sadly common practrice of clipping the ground pin from the mains power plug of the device, deleting the safety ground. A separate wire is then run to the audio ground — possibly the shield or drain wire of a signal cable. This is a flagrant code violation, and is *extremely* dangerous. The ground impedance of such an arrangement may well be inadequate to sustain fault current, creating a major life hazard.



An entirely safe audio system which is code approvable may be enjoyed when the audio ground scheme is derived by way of the required equipment safety ground conductor. Due to the resultant low ground impedance — the ground conductor is sized by code to handle fault current — audio performance is often considerably superior to that expected of a traditional, and possibly unsafe, ground scheme.

The trick, if it be called that, lies in the design of the mains power system: *each branch circuit includes conductors for hot* (black or other color), *neutral* (white) *and equipment ground* (green), *of equal cross section area, continuous from receptacle to panelboard.* Notice that this scheme specifically excludes the use of single-neutral multiwire branch circuits.

As equipment ground bus is provided, bonded to the panelboard; this is in turn grounded via the mains supply raceway (conduit). By code, the neutral ("identified conductor") also must be bonded to ground at the service entrance point. The hot-neutral-ground conductor arrangement of the suggested scheme is intended to maintain a



neutral impedance approximately equal to the ground impedance under conditions of varying load — such as would be imposed by a large console or multiple power amplifiers. Please see Figure 4.

The straightforward application of the scheme requires that all mains power circuits be fed from the same neutral bus. The practical way to do this is to derive from a single panelboard all mains circuits feeding the sound system. The more spread out the system, the more important is this requirement, as the ground impedance will increase, resulting in higher possible ground voltage differences between items of equipment, thus greater likelihood of induced noise.

In the event that the raceway derived ground at the panelboard is expected to be excessively noisy, a further refinement may be added to the preceding scheme. This is detailed in Article 250-7A, Exception 4, of the National Electric Code (1978 edition).

Exception No. 4: Where required for the reduction of electrical noise (electromagnetic interference) on the grounding circuit, a receptacle in which the grounding terminal is purposely insulated from the receptacle mounting means shall be permitted. The receptacle grounding terminal shall be grounded by an insulated equipment grounding conductor run with the circuit conductors. This grounding conductor shall be permitted to pass through one or more panelboards without connection to the panelboard grounding terminal as permitted in Section 384-27 Exception, so as to terminate directly at the applicable derived system or service grounding terminal.

Some caution in following this scheme is necessary. The National Electrical Code is the model for most wiring practice in the United States, and is adopted in its entirety by many jurisdictions. It is best to verify that this scheme is acceptable to the local authorities having jurisdiction. As an example, the above clause is included in the Electrical Code of the *County* of Los Angeles, it does not appear in the current code of the *City* of Los Angeles.

This approach illustrated in Figure 5 creates a "clean" or "technical" ground free from ground noises due to odd ground currents in the raceway system. Notice that a *completely* separate ground system is permitted — ground rods in earth and all. The "clean" ground scheme has been found effective for extremely large sound systems, demanding low-noise requirements, and areas with otherwise difficult radio frequency interference (RFI).

All receptacles providing audio system mains power should be tested for correct polarity of connection and ground continuity before energizing the audio equipment. A surprising number of receptacles are found mis-wired in the field.

All signal shields should be insulated from conduit and enclosures. To maintain the unipoint ground scheme, it will also be necessary to insulate microphone connector shells (XLR-3 series compatible) from outlet boxes and raceways — stock connector plates may not provide this essential feature.

If the connector shells of portable audio cables are of conductive material, it has been found wise to insulate the shell or float the shell with respect to ground. This will prevent the creation of intermittent ground loops due to contact of connector shells with those of another cable or other grounded surfaces, such as outlet box covers, conduits, and the like. This points up another general rule — *bond firmly to ground or insulate from ground* — casual or intermittent grounds will make troubleshooting a nightmare.

It is a paradox, but is actually easier to establish a rational and safe ground scheme for a portable touring sound system than for a permanent installation. In the touring situation, you will generally carry your own power system; with three-wire grounded portable cords and your own distribution panelboard, system ground paths are entirely predictable — it is only necessary to find an earth connection in the form of a rod or cold water pipe.

In a pinch, a safe and functional portable power distribution system may be rented. In most jurisdictions, construction power distribution systems approved for use in wet excavations will meet all the foregoing standards and are generally equipped with Ground Fault Current Interrupters (GFCI). These are modified circuit breakers which will sense and trip out on a ground fault current condition, a handy safety feature when you happen to be playing an outdoor gig in the rain.

Caveat

As mentioned earlier, the proposed method for combining mains power and audio grounds is but one valid method for achieving a functional, reliable ground scheme. Admittedly, it goes against much conventional practice in the field. It is not likely to be a panacea; our readers may well be able to discover situations where other approaches will better suit their needs.

Lest the reader be left with the impression that this is a recent fantasy of the author, some practical experience in application should be mentioned. The author has been using the scheme in work specified in his consulting practice since 1972; it was developed at that time to deal with the problems presented by five complex, interconnected control rooms for a university research laboratory. Since then, the approach has been used for recording control rooms, discotheques, high-level entertainment sound and live theater sound effects systems. There is no claim that the presented scheme will work every time; however, it has proven effective every time it has been used thus far.

Acknowledgements

The author wishes to thank Peter Butt, Bill Isenberg, Deane Jensen and Dick Rosmini for their comments and suggestions during many discussions of the topic.

References:

For further discussion, and in some cases, alternate opinion, the interested reader may wish to consult the following:

Kevin Cousineau, *"Grounding and Shielding Techniques for Portable Concert Reinforcement Systems"*, preprint No. 1259, Paper D-7, presented at the 57th Convention of the AES, May 10-13, 1977, Los Angeles.

Don Foster, *"Grounding!"* Recording Engineer/Producer, Vol. 4, No. 2, March/April 1973.

Ralph Morrison, *"Grounding and Shielding Techniques in Instrumentation"*, John Wiley & Sons, Inc., 1967.

National Fire Protection Association, *"National Electrical Code"*, *1978 Edition*, NFPA, No. 70 - 1978.

Henry W. Ott, *"Noise Reduction Techniques in Electronic Systems"*, John Wiley & Sons, Inc., 1976.

"Recommended Wiring Practices", RCA Broadcast Equipment Catalog.

"Electronic Systems Wiring and Cabling", Trompeter Electronics, Inc., Catalog T-10.

ACOUSTIC & ARCHITECTURAL DESIGN OF STUDIOS:

the existing structure:

SOUNDMIXERS In The Heart of N. Y. C.

Design and construction of studios in existing structures almost always imposes some acoustic and creative compromises, if not outright restrictions, on architects and acoustical designers. Such was decidedly the case in surveying the fourth floor of New York City's historic Brill Building in the heart of Tin Pan Alley for use as a four studio recording complex.

The problems as they generally exist in converting typical office building floor space into recording studios, and specifically in high-rent mid-town Manhattan, are two-fold: 1) Obviously wanting to make the most economically efficient use of the space, and 2) accommodating the acoustic design to the low ceilings which generally exist in such buildings. These two design objectives are closely related.

In addressing the ceiling problem, here are three basic approaches that can be affected: First, after the raw acoustic treatment has been applied to the ceiling, the service systems, air conditioning ducts and electrical apparatus can be fastened directly to the ceiling and left exposed. In most cases this approach, although it conserves most of the needed vertical space, is less than satisfactory. It is also, contrary to most opinion, not always the least expensive way to install the systems because of the need to use more expensive materials.

The second method is to install everything up on the ceiling to then be covered by a *skin*. Where height is no problem this is certainly an acceptable method of ceiling construction.

SOUNDMIXERS' Studio A. *Note services installed in perimeter soffit, preserving maximum vertical dimension. Photography by* **Robert Wolsch**

Control Room Design Specifically for the Small Studio

by
Woody Smith
Abadon/Sun
San Antonio, Texas

The articles previously published in *Recording Engineer/Producer (April, 1977 and October, 1977 Letters to the Editor)* attempted to outline some of the fundamental considerations that ought to be taken into account in designing any control room. This article will continue to develop those ideas somewhat and look at some new methods for determining the quality of a room design.

First, to summarize the information contained in the previous articles:

A totally bounded structure will exhibit a series of resonances when excited by a coherant signal such as a loudspeaker reproducing music. In an untreated rectangular room these resonances will be directly related to the room's dimensions, and will occur as multiples of the fundamental room in a given room mode. (Note: In the real world these room resonant modes are dependent not only on the room dimensions but also on wall construction, room shape, acoustic treatment, loudspeaker placement and the presence of equipment or personnel in the room. Nonetheless, we feel they're a very important part of the initial design aspect of a control room.) These resonances occur throughout the audio spectrum but are of interest to us particularly at low frequencies where the effects are most audible. At higher frequencies the proximity of resonant frequencies to each other, and their very narrow bandwidth, make them inaudible in a majority of instances. The calculations given below are used as a

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Control Room Design for Small Studios —

basis for determining a set of room dimensions which will exhibit the least number of adjacent or overlapping resonances. (The presence of multiple overlapping resonances will tend to produce very irregular low frequency response.)

$$f_n = \frac{565n}{d}$$

where n = 1, 2, 3, . . . ; f = res. frequency in Hz; and d = room dimension in feet.

For example, a room measuring 10′ x 12′ x 15′ has theoretical resonances occuring as follows:

f_n	10′	12′	15′
f_1	56.5	47.1	37.7
f_2	[113.0]	94.2	75.3
f_3	169.5	141.3	[113.0]
f_4	[226.0]	[188.3]	150.7
f_5	[282.5]	235.4	[188.3]
f_6	339.0	[282.5]	[225.0]
f_7	395.5	329.6	263.7

Those figures which have been bracketed above are resonant frequencies which are common to more than one room mode. In this case, our room dimensions have yielded a high number of overlapping resonant frequencies.

Evaluating a different set of dimensions:

f_n	9′	11′	14′
f_1	62.8	51.4	40.4
f_2	125.6	102.7	80.7
f_3	188.3	154.1	121.1
f_4	251.1	205.5	161.4
f_5	313.9	256.8	201.8
f_6	376.7	308.2	242.1
f_7	439.4	359.5	282.5

As you can see, a choice of dimensions lacking a common divisor also lacks common resonances..

The ratio of dimensions selected should also meet an additional qualification in that it should lie within the boundries given in the graph below.

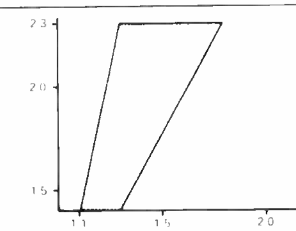


Figure 2 RANGE OF ACCEPTABLE RATIOS

Another calculation that can be made at this point relates to the potential low frequency response within a room. This is a function of the room directly related to it's volume and longest dimension, where a larger room will have a more extended low frequency response capability.

$$f_0 = \frac{565}{\sqrt{w^2 + l^2 + h^2}}$$

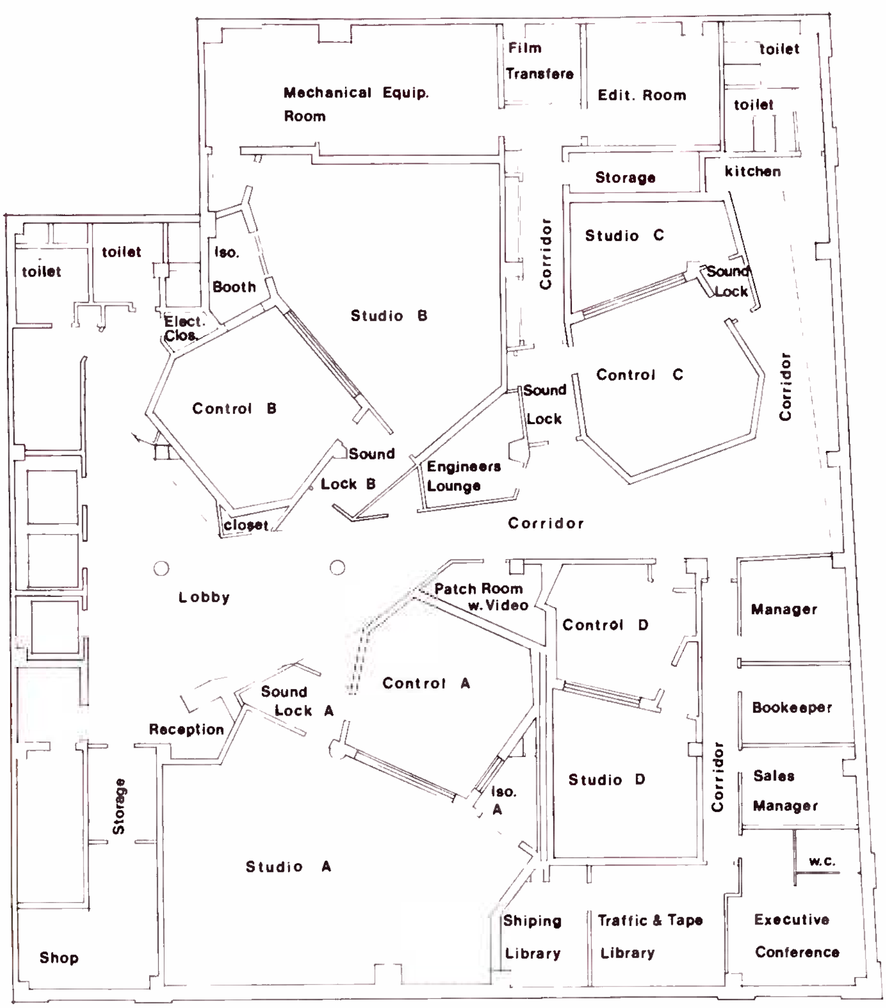
where f_0 is the frequency of the 1/2 wavelength dimension, w, 1, h are room dimensions in feet.

The overall room volume should be selected to allow at least for 35 Hz ½ wavelength propagation. If a larger area is available, it should be used as the bass reproduction will become audibly more "solid" with increased room size.

SOUNDMIXERS — *continued*

The Soffit System

The third method, called the Soffit system, as was most effectively employed in its most advanced form in the Soundmixer's project, consists of mounting all of the mechanical services on the perimeter of the ceiling, then covering them up. This leaves a large open expanse of high ceiling in the middle of the room. By constructing in such a manner a soffit system was created which is exposed and accessible, but at the same time still gives the designer the opportunity to employ a goodly amount of tricky acoustic geometry. The wooden *cloud* system suspended from the ceiling, as shown, accentuates the feeling of height and space in the room.



All these considerations are important in forming the theoretical fundamentals for a control room design.

Pratical Design Considerations

All this sounds very nice, but . . .

The two major problems we've enountered in the design of small control rooms are severe peak to trough amplitudes in swept-sine-wave frequency response and "acoustic overload", i.e., where the amplitude of reflected signals in the room is nearly equal to the incident signal from the monitors.

Although these two problems sound quite different they stem from a common problem — the small size of the room. This forces us to use a somewhat unique design approach whenever size limitations are a factor. Let's consider the physics of sound in a small bounded enclosure (a rather dramatic way of saying a small control room).

Sound level decays in proportion to its distance from the source. As the distance increases between the monitors and the reflective room surface under consideration (the rear wall for example) the amplitude of the reflected signal will diminish. Unfortunately, in small control rooms the distance between the monitors and rear wall isn't sufficient to provide the necessary decrease in amplitude of these reflections. When a sound wave bounces off the rear wall and heads back toward the console it arrives with a certain amplitude and some degree of phase shift, determined by the distance from the console to the rear wall and the frequency. Certain frequencies will have reflections which will arrive at the console in phase with the direct monitor signal and add, while others will arrive out of phase and produce nulls. The same situation occurs with reflections off the side walls of the room. As the amplitude of reflected signals increases their ability to alter the measured response drastically increases, producing severe peaks, nulls and a generally *'crummy'* response.

The effects of high level reflections, although very audible in a room, are not readily apparent when using conventional 1/3 octave room analysis. The peaks and nulls produced by these interactions between direct and reflected signals are very narrow band phenomenon and are concealed by the averaging of levels that occurs in 1/3 octave analysis. It may actually result in a band being erroneously equalized to compensate for a severe peak or null. Although these peaks and nulls *are* measurable using pink noise, the bandwidth must be reduced to 1/6 to 1/10 octave and very slow plotting speeds used (32-64 sec.). Even under these conditions, the frequencies below 100 Hz are questionable due to the random nature of the noise source. To provide increased sensitivity to these room anomalies, we use a swept sine-wave room analysis.

The first criterion for evaluation of a room is the amplitude and quantity of these anomalies. The lower the amplitude and fewer the number of deviations, the better. Since these peaks and nulls are location dependent, and are not equalizable, they must be identified and then deleted for room equalization purposes.

By this method, we can equalize the overall (majority) room response independent of localized narrow-band resonant effects. One further benefit of using swept sine-wave analysis of a control room is the correlation that exists between measured sine-wave response (before EQ) and room reverb time. When a room exhibits resonant modes and other steady-state interactions, there will usually occur some direct relation between the amplitudes of these interactions and the decay time of a signal in the room. The peaks and nulls indicated by the unequalized s/w response plot correspond to frequencies at which positive or negative interactions occur between wavefronts. Positive interactions will tend to minimize signal loss, resulting in longer decay times while negative interactions will produce shorter decay times. These correlations will hold true in a majority of cases, although rooms exhibiting RT_{60}'s with distinct decay discontinuities are not totally predictable.

Nonetheless, we can say that a room which has a well-behaved s/w response will tend to exhibit a resonably uniform reverb time.

The other phenomenon present in small control rooms is "acoustic overload". At high monitoring levels in small rooms a point is reached at which the room seems to overload. This really isn't the room being unable to handle the acoustic load but the listener being bombarded by a cocophony of direct, high-amplitude reflected and reverberant signals. The short time span between the initial incident sound from the monitors and the reflected sound drastically reduces the intelligibility and

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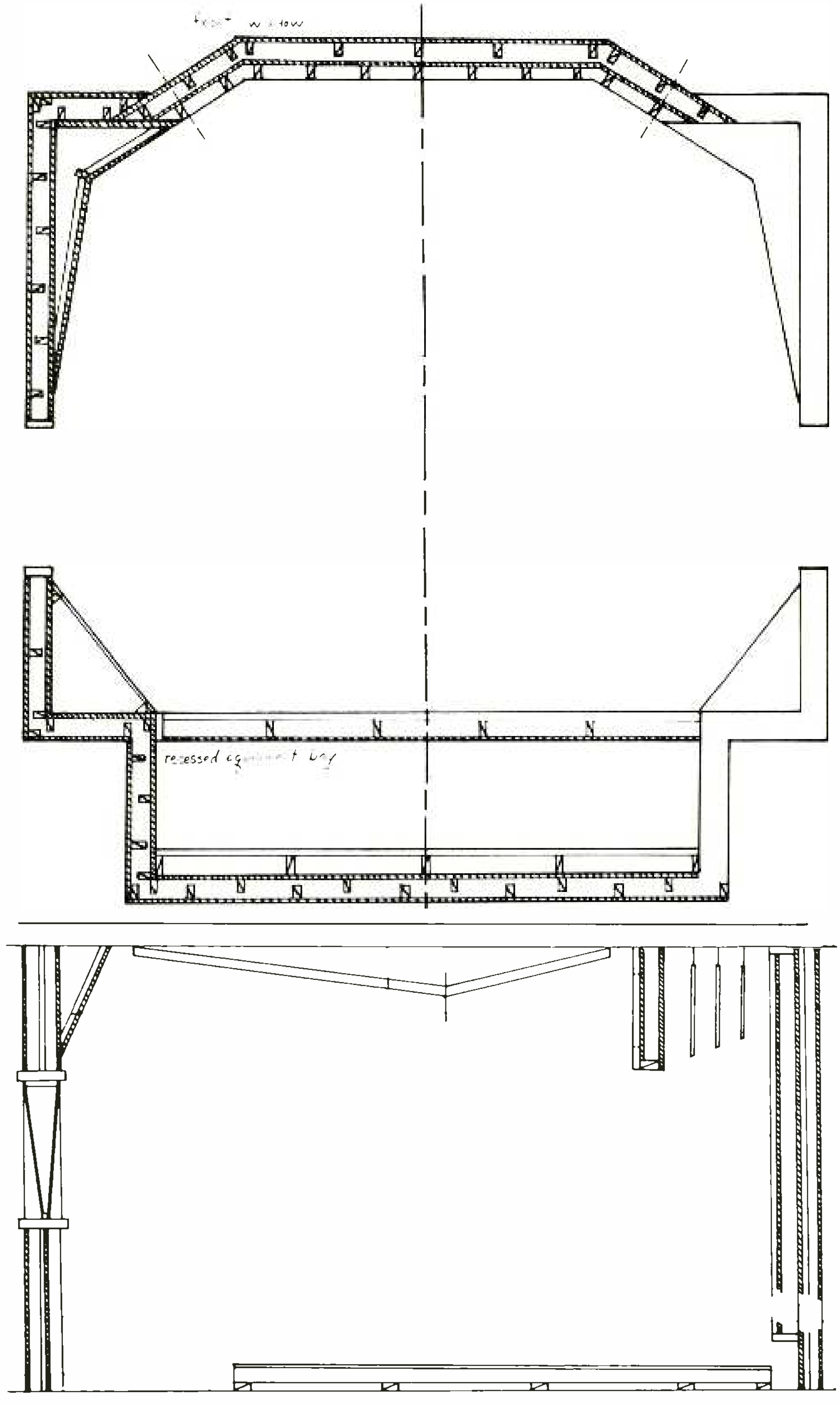


clarity resulting in a "mud factor". Added to this is the room's own reverberant charactertisitics, resulting in an accumulation of sound within the room to the extent that the delayed and reflected signals may become nearly as loud as the direct sound from the monitors.

Correcting the Problem

Now that we've identified the problem, how do we correct it? If we totally deadened the room we wouldn't have the problem any longer but the room wouldn't sound natural either. So, we must locate the best surfaces for absorbancy and treat them to the extent that the proper decay time, as determined by the reflection amplitudes, is reached while retaining a relatively flat room response. Achieving the proper decay time using treatment while retaining flat response (i.e., linear frequency vs. reverb time and linear frequency vs. amplitude response) isn't that easy. So, it becomes the basis for our small control room designs. By altering boundry absortion we can alter the room's reverb time, as if we'd altered the rooms size, eliminating many of the problems inherent in its limited physical dimensions. Again, one very important consideration is that the attenuation provided for the purpose of altering reflection amplitude must have a linear absorption vs. frequency characteristic or it may result in severe frequency response and reverb time anomalies.

Figure 1 shows a recent control room design by us (ASI). This design emphasizes an increasing room cross-sectional area in the horizontal and vertical planes to provide for an actual decrease in net SPL with distance from the monitors. The vertical dimension takes on an increasing height aspect while on the ground plane the narrow front section is widened toward the center of the room. This results in the monitors radiating from a smaller cross-sectional area into a larger one. The dispersive aspect of the room achieved in this way provides for a more even SPL throughout the room with reduced susceptibility to "overload". Another room treatment utilized is an active ceiling, which serves to reduce reflections off the ceiling into the vertical plane of the room. Controlled horizontal dispersion is very important in achieving a proper "room sound" and natural room reverberation, but vertical plane reflections in a small room, while not adding to the quality of sound, can result in a measurable deterioration of response within the room. Additionally, the treatment of the ceiling further increases the "acoustic dimensions" of the room, within the operational limits of the treatment used, resulting in a more uniform response throughout the spectrum. The front section of the room is constructed from hardwoods and rock to provide surfaces which are irregular and dispersive at high frequencies. The angled sections are formed by a wall of 2 x 4's on 16" centers covered with insulation sheathing (highly damped) and the appropriate wood covering firmly attached to the sheathing to eliminate any possiblity of rattles. The cavity formed behind this angled wall is filled with fiberglass insulation or acoustic foam. Also, the area below the control room window is covered by a trap packed with insulation and covered with cloth. One thing to note regarding the placement of bass

Figure 3A

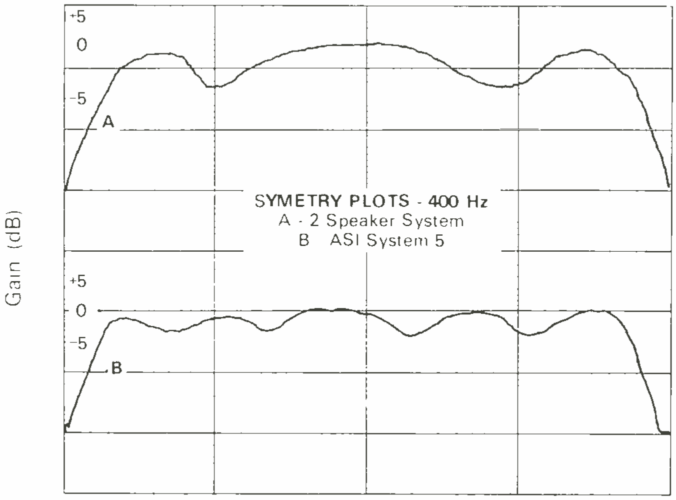
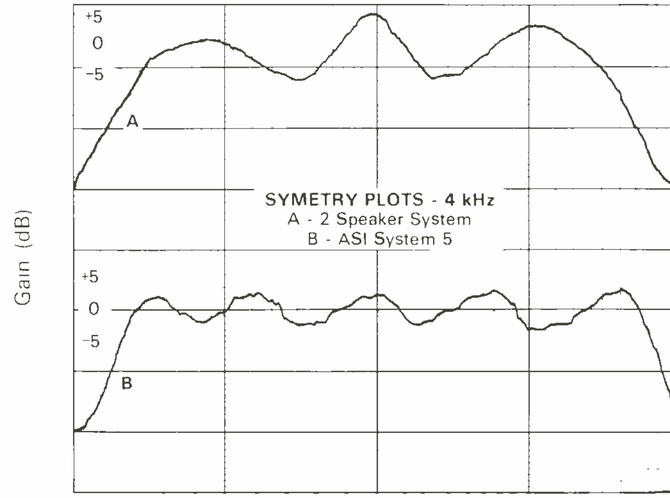


Figure 3B



trapping in particular is the omnidirectional nature of sound at frequencies below 100 Hz. This allows for placement of bass traps in locations which are convenient to the layout of the room and out of direct sight. You will find, however, that the greatest concentration of bass energy in a room will occur at junctions of room boundries, i.e., corners. The concentration of bass in these locations lends them to being ideal locations for bass trapping when the usable space is very limited. You will note on the side-view of the control room drawing in Figure 2 that the high ceiling area inside the recessed equipment bay at the back of the room has been used for trapping, particularly of low-frequency components. Also, the rear wall of the equipment bay is used as a mid-frequency absorber as is the upper wall section above the bay. The two angled corner sections of the room toward the back are also bass and mid/high frequency traps. The trapping of the rear section of the room in this fashion increases the allowable direct monitor output while reducing the amplitude of rear wall reflections, thereby lessening the "overload" tendencies and improving stereo imaging.

One point that has previously been made is the need for absolute symetry in the control room. The reasoning behind this is that the effect of reflected sound is not only measurable but very significant in determining the sound as it will be heard at the console. A lack of symetry may result in a vague stereo image for an image which "moves" with frequency. Shown in Figure 3 is a plot of room symetry in which a microphone is moved across a room and the amplitude of sound plotted as a function of location. By performing this test at several frequencies, any symetry errors can be detected and the useable monitoring area determined (we consider useable monitoring area to be the area within the -3 dB points to either side of center). The three peaks occuring in the plot represent the values of the left, summed (mono) and right signal levels. It is important that a strong mono image be present to allow for a smooth transition as a signal is panned from left to right and vice-versa. The amplitude and location of the center image is determined in part by side wall reflections. As a result, a difference in wall treatments or physical room asymetry will alter the imaging. This symetry problem is most acute when two monitors are in close proximity to the monitoring area, as they are in a small control room. It becomes very difficult to have a wide stereo image while retaining a strong center image and smooth left to right

Figure 4A

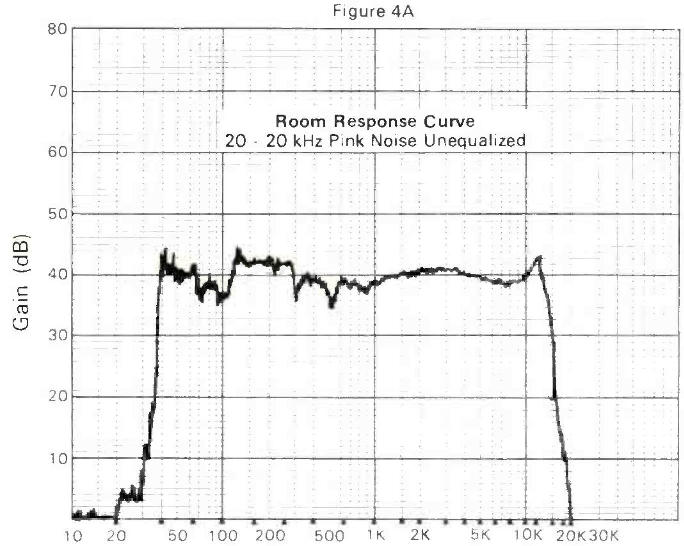
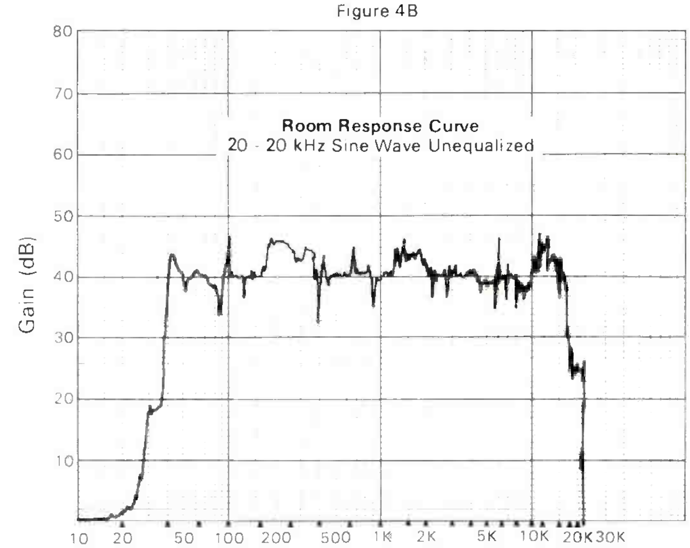


Figure 4B



(or right to left) panning. To help aleviate this problem, a multiple speaker system can be utilized if combined with the proper electronic tailoring. In a multiple speaker system the "stereo image" is reproduced by a series of speakers across the monitoring plane, allowing for a very wide stereo spread while retaining amplitude and location information for all points in between. This system can also reduce the level deviation occuring on either side of the center, thereby widening the acceptable monitoring area and providing a more linear response throughout the control room.

Hopefully, this will have provided you with a little better understanding of what we're doing with small control room design. Figures 4A, B and C have been included to give you an idea of the frequency response capabilities of a room similar to the one discussed here, while Figures 3 and 3A compare the symetry plots derived using a two speaker monitor system and our own ASI System V monitor system using multiple speakers. All the plots and measurements contained here were made using an Amber 4400A Test Set, with all comparative testing done simultaneously to eliminate any errors.

Note: Reprints of the original article are available from ASI, P. O. Box 6520, San Antonio, Texas 78209.

A Control Room/Studio Communications System

by
Bob Todrank
and
Gary Carrelli

Most studio owners have had occasion to need some piece of equipment which is not currently available from professional manufacturing companies. When this type of need arises, one must either modify existing equipment to approximate what is necessary or custom build a unit to fit the exact requirements. One such unique situation arose at Bee Jay Recording, in Orlando, Florida.

Since Valley Audio served as consultant on this new studio project, we had to become acutely aware of Bee Jay's unique requirements. Throughout our initial meetings the necessity for "total" communication between Control Room and studio became apparent. This was a major concern for the principals involved because of past experience in their other studios. The need for extremely flexible communication in the new Studio "A" was accentuated by the following:

First: The large size of Studio "A" made complete visual communication between Studio and Control Room difficult in some areas. This size also lends itself to accommodating large numbers of musicians on the same session, which also confuses C. R. — Studio communication.

Second: Bee Jay uses an in-house arranger/conductor on many of its sessions. In most cases, he would be directing a large string section overdub or group of backup singers, etc. His main complaint about their previous studio was a problem in direct and immediate communication between himself, the engineer, and the musicians. The lack of communication seemed to be the main contributor to slow sessions, confusing instructions, numerous mistakes, and loss of valuable studio recording time.

Third: The superior acoustic isolation between C.R. and Studio "A" made the all too often "yell through the C.R. window real loud" totally impossible.

Definition of Requirements:

The development of a useable communication system came about in numerous stages. Our initial ideas were expanded many times before completion of the final hardware. Bee Jay's request was for a "total aural communication system" between C. R. and Studio. One with enough flexibility to cover all the requirements yet compact and easy to use. Some specific requirements were outlined.

1. The system must be extremely mobile so that a conductor could move it quickly and easily anywhere in the studio.
2. Since the conductor/arranger also plays piano on many sessions, the unit must be usable at the piano while also being useable in the large string room with the door closed.
3. The system must be reliable.
4. The control package must be as small and lightweight as possible.

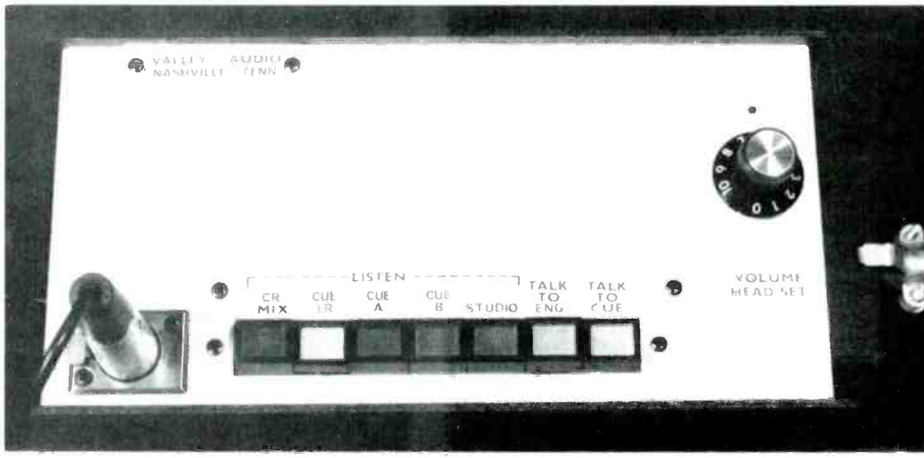
With these ideas in mind, we set out to design and build a system around the given parameters.

Our first decision was that of basic format and layout. We decided on essentially two control stations, one at the console and one remoted in the studio. For the sake of appearance, operation, and ease of hookup we decided to build the console station into a blank panel in the face of the mixing console, a Sphere Eclipse "C". Sphere Electronics was kind enough to mill one of the blank panels in the console to our specifications, and mount the necessary surface parts (5" speaker, function select buttons, speaker level control).

We decided to build the studio control station into a small box that could be placed on a music stand or piano bench. This box was to terminate in a multi-pin connector so as to be removable from all cabling for storage. Two 25' multi/pair extension cables were to be used to connect the box to

another multi-pin connector mouted under the C.R. window. To make the unit useable for string sessions, we decided to parallel connectors and place a duplicate inside the string room. The two cable lengths could now be hooked together if required, to run most anywhere in the studio while only one section was required to use the system in the string room.

Our second decision was that of overall function. If we were to gain two-way communication between C.R. and Studio, we obviously required a mike and speaker in both locations. The obvious choice in the studio was a stereo headset with attached microphone. As far as the C.R. was concerned, we had to decide whether to use the C.R. monitors as communication speakers, or install a separate speaker for this purpose. We decided to go with a small speaker just for communications so as not to confuse the engineer with a conductor talking over his monitor mix. The normal console talkback microphones were to be used for engineer talkback to conductor.



. . . the studio box showing available functions.

Our third decision was to define specific detailed functions:

1. Conductor must be able to talk to the engineer at anytime he wishes. How many times have you waved your arms and screamed in the studio trying to attract the attention of an engineer?

2. Conductor must be able to talk to all console cue busses so he can communicate with musicians wearing their headphones during a session.

3. Conductor must be able to monitor in his headphones any combination of the following sources at the push of a button. This must be totally independent of what is taking place in the Control Room.

a) C. R. monitor mix — The stereo monitor mix out of the console regardless of whether this mix is being monitored in the C. R.

b) Stereo Cue Buss — Monitors output of consoles stereo cue buss.

c) Cue A — Monitors output of console mono Cue A buss.

d) Cue B — Monitors output of console mono Cue B buss.

e) Ceiling Mike — Monitors output of a mike suspended high in the center of the studio. Through this mike, the conductor can monitor anyone speaking anywhere in the studio without removing his headphones. This is necessary especially to pickup those people in the studio who are run direct and have no close mike feeding the cues.

4. The engineer must be able to monitor the suspended studio ceiling microphone at the push of a button. We felt that this function was extremely important. There are many times when setting up for a session or cutting a direct overdub, that someone tries to talk to the engineer from the studio. The engineer frantically tries to locate an open mike, assign it to a buss, monitor it, and — oops, too late, the conversation is over. The engineer can now just touch one button and instantly hear anyone in the studio through his small communication speaker, anytime he wishes.

5. The engineer must also be able to feed the console cue systems with the overhead mike. This covers the time when a guitar player, overdubbing through a direct box, wishes to speak to the synthesizer player who is across the room and wearing headphones.

6. The engineer must be able to talk directly to the conductor's headphones without disturbing anyone else.

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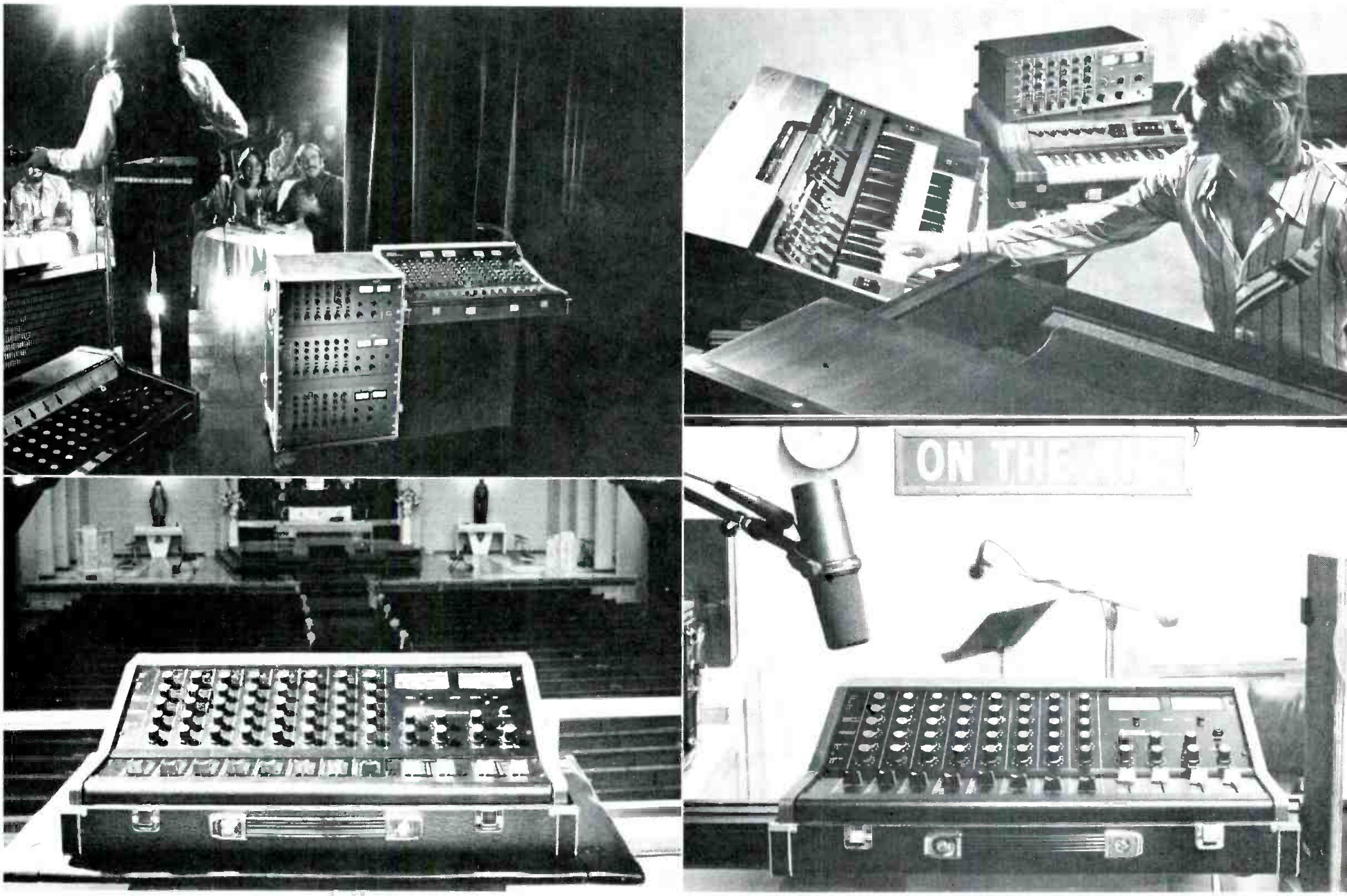

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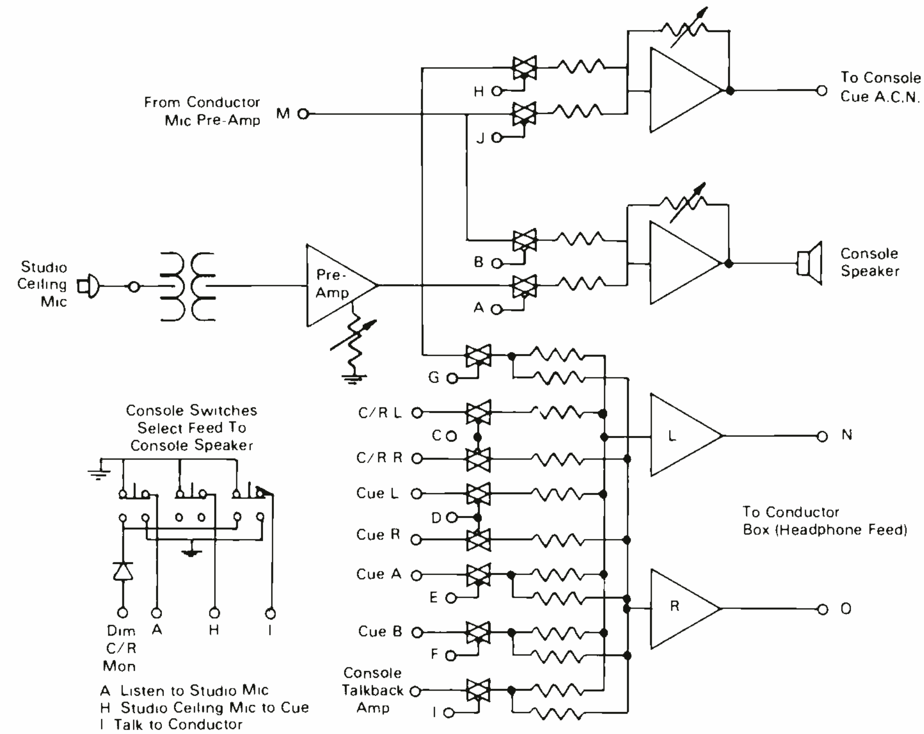
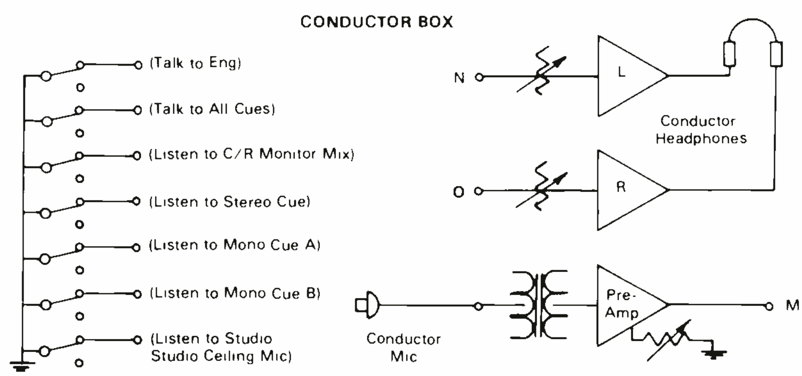


CONDUCTOR BOX



Final Electronic Design Considerations:

Before heating up the soldering iron, the system approach and limitations had to be considered. For instance, since we were tapping console power (±24 V) and needed ±15 V for our circuitry, regulation was necessary. To avoid excessive size and heat dissipation in the regulator, low power consumption was of major importance.

The quality of all audio signals had to be excellent to provide low noise, highly intelligible communications from studio to Control Room. Careful gain structuring to avoid clipping and excessive noise must be adhered to throughout the entire system. Foolproof operation and well labeled, easily understandable controls were required for hectic sessions and for the benefit of visiting engineers and producers. And last but not least, that old project killer, it had to be cheap!

Electronic Construction:

The system concept was basically as follows: One major circuit card, located at the console, would provide all necessary switching and console buffering or interfacing. System power would be locally regulated (on card) and distributed with audio and switching voltages through a multicable to the remote conductor box. The console would have a small power amplifier, running off unregulated ±24 V, driving the five-inch speaker built into the front panel of the console. Speaker volume as well as talk and listen functions would be easily accessible to the engineer and all communications system functions would utilize the auxilliary five-inch speaker. All levels to and from the conductor box would be +4 dBv.

The conductor box included a stereo headphone amplifier — 4 watts per channel; a microphone pre-amp which boosted the conductor mike level to +4

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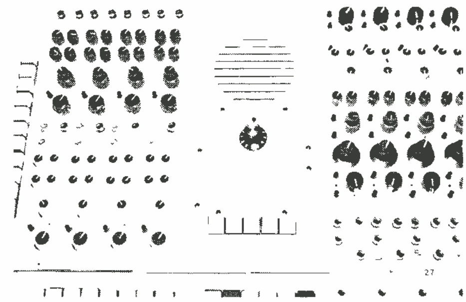
dBv; and a set of switches for the conductor's talk and listen functions (previously described).

For the conductor's headset we chose a Sennheiser HMD-224 for its superior headphone isolation to and from the outside world, and the excellent microphone response curves. The conductor mike was coupled via a transformer (no Trans-Amp LZ's back then) into a LM 382 pre-amp chip located in the conductor box. The LM 382 was chosen for its minimum parts count and coinciding voltage requirements. Flat frequency response was maintained in the pre-amp because of the low end roll-off incorporated in the mike. Proximity effect was therefore minimized in the mike itself. Thirty to 40 dB of gain was all that was necessary because of relatively high levels provided by the mike's close proximity to the mouth.

Selection switches for conductor talk and listen functions were momentary and push-on, push-off respectively. The listen functions were capable of all being depressed simultaneously for multiple source monitoring. (i.e., Cue A on top of Cue L & R.) Again these switches were merely controlling D. C. levels which turned analog switches on and off at the console (Example: The conductor punches up Control Room mix; a D. C. level is provided on a predetermined wire which turns on the pre-fader Control Room mix analog switches. These two signals, left and right, are then summed into their respective combining amplifiers, boosted to +4 dBv and returned via the connecting multicable to the conductor box. Here it is amplified [power amplification] and appears in the conductor's headphones.) Since the only voltages present in the interconnecting multicable were ±15 V, D. C. low current switching, and +4 dBv audio signals, possibilities of oscillation and crosstalk from level mixing were minimized.

At the console end, as mentioned earlier, CMOS analog switches were utilized. The reasons were fourfold. *1.* They draw miniscule currents (approximately 40 microamps per 4 switches); *2.* They provide noiseless D. C. controlled switching; *3.* They are compact (four SPST switches per 14-pin dip package); and *4.* They are inexpensive (approximately $1.00 for four switches)! These switches,



— the control room connection — installed in Bee Jay's Sphere Eclipse C console

however, must be used with care. Their maximum input is +14 dBv, period! The healthy +24 dBv outputs of the Sphere had to be padded at the input of the analog switches. This proved to work well however, because we recovered the level in following stages with negligible loss in S/N.

Since the engineers talk control that we added (talk to conductor) indeed opened his console talkback mike into the conductor's headphones, we decided it was necessary to dim the Control Room monitors just as the console did for its talk functions. An interface to the monitor dim circuitry had to be made to the Sphere. This turned out to be extremely easy because of Sphere's standard grounding logic for this function. We merely grounded the dim buss through our "talk to conductor" switch.

The ceiling mike was to be carefully chosen to fullfil all of the demanding requirements of the system. We, however, ended up using an EV-635A because it was "the only mike they had laying around"! Fortunately, because of its reasonable sensitivity and good omnidirectional characteristics, it was quite suitable for the application. The ceiling mike cable was brought up to the console, bundled with all the other mikes, and routed into the main communications circuit board. Here it was coupled through a transformer and into a LM 381A pre-amplifier IC. This chip is very similar to the LM 382 except for its extra pins giving more elaborate gain control. Also, two of these pins allow you access to the input stage of the amplifier. To squeeze a few more dB of signal-to-noise from it, we increased the current density in these input stage transistors. This amplifier ran at about 40 to 50 dB of gain (trimmer adjustable) to provide a reasonable signal level with normal studio acoustic levels. The gain trim allowed us to match its output into the console speaker summing amplifier to obtain matched levels with the other monitored sources (conductor mike).

The console five-inch speaker was driven with a small four watt push-pull amplifier of our own design (similar to the two driving the conductor's headphones), and fitting with a 6 dB panel mounted gain trim control. The trim feature was utilized to avoid maddening attempts on the conductor's part to attract the attention of the engineer who inadvertantly turned off "that little speaker that kept yelling at him". (We must be conductors at heart.)

Problems:

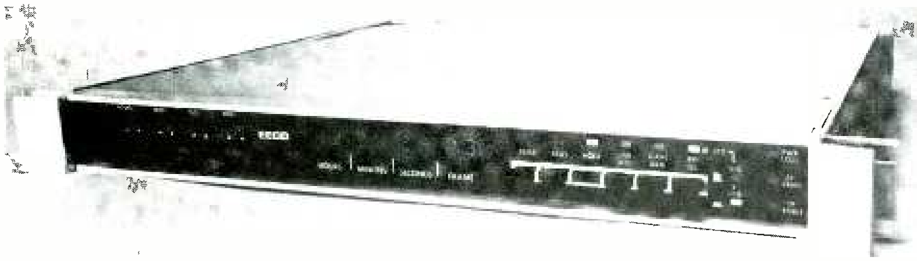
A few problems, as expected, were encountered upon installation. With the number of sources being summed and the multitude of panel accessible knobs in the system, there were bound to be problems with level matching. Many of the pickoffs for the conductor listen functions (namely Cues) were post their master faders. This required that the master knobs remain fixed and careful engineering was necessary to maintain constant cue send levels. Both were proven to be inconsequential because of a good engineering staff and a minimum of knob twiddling by extraneous people.

Since both engineer and conductor could talk to each other simultaneously, some care had to be taken to adjust levels properly to avoid feedback problems.

There was one problem that proved quite embarassing, however. It seemed that the input strip to the immediate right of the communications panel (out speaker panel) wasn't working properly in all modes. After several hair pulling troubleshooting sessions, it was discovered that the speaker magnet was holding open a few key reed relays in the adjacent module. (Ooooooppps!) Well, nothing that a small plate of metal wouldn't cure.

Conclusion:

Considering all, the system was a giant success. It is being used daily at Bee Jay Recording. It is also interesting to note that during single overdub sessions, when the instrument is being taken direct, the musician likes to simply clamp on the conductor's headset and use the full communications capability with the engineer. Personally we believe the system is a success because of its capability to improve personal communcations (always a help), and a lot of that good old red blooded need to push a lot of buttons!



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NEW SPECTRA SONICS MODEL 404RS POWER SUPPLY

A new Spectra Sonics power supply, the Model 404RS, is now in full production. The supply is the culmination of a research effort to produce a DC power supply fully capable of withstanding the hard use that is a part of "on the road" tours, with no maintenance.

The power supply is manufactured in two configurations. One Model 404RS provides 48 VDC (plus and minus 24 VDC) at 8 amperes. The other, the Model 404 RSD (Dual) can provide 16

amperes, bi-polar DC.

A heavily constructed steel chassis of the same physical size (5¼" high x 19" wide x 6" deep) for installation in a standard electronic equipment rack is used in both configurations.

A Model 404RS ($215.00) and the Model 404RSD ($410.00) are "on the shelf" for immediate delivery.

SPECTRA SONICS
3750 AIRPORT ROAD
OGDEN, UT 84403
(801) 392-7531

for additional information circle no. 65

DELTA DL-1 DIGITAL DELAY

Designed for professional and semi-professional recording studio and for sound reinforcement applications, the unit features:

Three outputs with independently selectable delays ranging from 5 to 160 ms.

If listening is your profession...

we've got some pros you should meet.

If you spend long, strenuous hours listening to high-level sound, you can't afford to be fatigued by distortion or distracted by poor quality audio. You've got to have the best low-distortion monitors with top power capacity and extended bandwidth that money can buy.

The answer—the professionals from Altec.

Recording studios worldwide rely on Altec studio monitors to deliver tight, crisp, accurate playback in the most crucial of professional audio environments.

As standards in the recording industry for over a quarter century, Altec monitors offer high efficiency, wide dynamic range and the distortion-free power required to help you make the right listening judgments.

For further information on Altec's line of studio monitors, including the Models 620/604-8G and 9849 shown above, please write to:
Altec Sound Products Division,
Commercial Sales Department,

1515 So. Manchester Ave., Anaheim, Calif. 92803 • 714/774-2900
ALTEC CORPORATION

ALTEC
the sound of experience

Frequency response of 30 to 15 kHz at all delay lengths and all outputs.

Dynamic range of 90 dB with THD of less than .2%.

The DL-1 also has both front and rear panel inputs and outputs permitting easily controllable feedback effects for studio or live performances.

For permanent sound reinforcement applications, internal tamper-proof controls are provided.

Suggested retail price is $1,200.00.

DELTALAB RESEARCH, INC.
25 DRUM HILL ROAD
CHELMSFORD, MS 01824
(617) 458-2545

for additional information circle no. 67

FOUR-CHANNEL OPEN REEL DECK

A four-channel open reel deck with Simul-Sync has been introduced by TEAC to succeed its 3340 model series. Designated the A-3440, the unit has three heads, three motors, and two speeds (15 and 7.5 ips) and features completely new transport, circuitry and logic boards. It accepts up to 10-inch reels and carries a suggested retail price of $1,500.00.

Bill Cawlfield, TEAC's director of product development, said that the new Simul-Sync mode is tied directly to the function selection position of each channel either for recording or overdubbing. The A-3440 accepts monophonic headphones and each channel can be monitored by simply pushing the appropriate switches. Optional pro curve dbx interface is available.

Microswitch, touch-button controls are provided and remote control is available. Other features include a manual cue level for fast search, cueing and editing, four VU level meters, mike/line input selectors, four front-panel mike jacks and independent level controls for each channel.

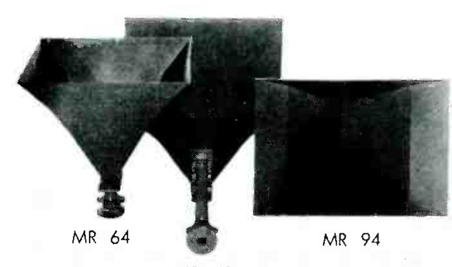
Specifications are listed as follows: 0.04% wow and flutter; 65 dB S/N ratio and 35 to 22,000 Hz frequency response at 15 ips.

TEAC CORPORATION
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
(213) 726-0303

for additional information circle no. 69

CONSTANT DIRECTIVITY CONTROL HORNS

The industry's need for a high frequency horn which would exhibit frequency-independent directivity control stimulated the development of Altec Lansing's Mantaray horns, according to Bob Davis, director of systems/applications engineering.



Altec is manufacturing the Mantaray in three different models — MR 94, MR 64, and MR 42 — that provides different coverage angles. The MR 94 offers 90 by 40 degree coverage, MR 64 gives 60 by 40 degree coverage and MR 42 provides 40 by 20 degree coverage. The horns are useable down to 500 Hz, but they provide best directivity control from 800 Hz to 16 kHz.

ALTEC CORPORATION
1515 S. MANCHESTER
ANAHEIM, CA 92803
(714) 774-2900

for additional information circle no. 70

HIGH PERFORMANCE GRAPHIC EQUALIZER

A passive-inductor, active-gain graphic equalizer designated Model RP2215-R is a "second generation" unit offered by Soundcraftsmen. Recently developed, computer-controlled coil

the ECLIPSE C

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Sphere is known for hard working, low maintenance console systems. The White House, the United States Senate, the Dorothy Chandler Pavilion are Sphere consoles . . . they *have* to work and they do!

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Sphere consoles have no built-in sound . . . they're "like a piece of wire *you* shape". "Transparent . . . clear . . . musical . . ." punch, with no "edgyness" common to IC consoles. We use discrete circuit amps at critical signal points — mike pre-amp, summing amps and output amps.

AUTOMATION
Sphere automation is designed to keep you on the air, not making excuses. We use the only *proven* system in the world . . . the Allison 65K. All others are hypothetical.

VERSATILITY
On each and every input we have three panable stereo sends *and* four mono sends (for echo, cue, effects . . . whatever you need 'em for), 9 VCA groups, 3 programmable mutes, 4 interchangeable equalizers (2 three-knobs, 2 nine-octave graphics and a four-knob *true* parametric available soon).

PRICE
You've heard about electronics that pass "DC to daylight" . . . well Sphere rolls off at the color green and puts it in your pocket. The Eclipse C is the finest console in the world for your dollar. It may be the finest console in the world for any number of dollars (or pounds or marks or yen). You're going to be very pleasantly surprised . . . we are priced very close to some of the mass-produced consoles presently available.

QUALITY
The Eclipse C is perhaps the most beautiful console you have ever seen. A truly elegant statement of the console maker's craftsmanship. Penny & Giles faders, Bournes rotary pots and the finest components available go into the 'C' . . . and a lot of T. L. C. as well.

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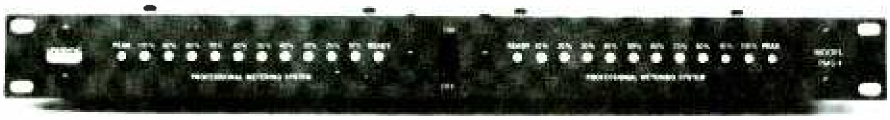
photograph taken at Bee Jay Studios, Orlando, Florida.
for additional information circle no. 71



winding equipment operating in conjunction with the precise setting of the center frequency of each of the 10 filter circuits to a specially designed Phase-Locked-Loop 10-band frequency generator enables an accuracy of center-frequency adjustment that the manufacturer says was not attainable with previous test equipment.

A 44 dB full boost/cut range is provided with each individual octave capable of ±22 dB when all octaves are set at full. A more meaningful measurement of ±15 dB per individual octave is guaranteed when all other octaves are set at zero, says the manufacturer. THD is less than 0.01% at two volts, signal-to-noise is 114 dB below full output, 100 dB at 2 volts and TIM is below the limits of the residual noise of the test equipment. Soundcraftsmen provides a rack-mount front panel as well as walnut-vinyl sides for attractive appearances on a shelf. With side panels removed, the 2215-R mounts on a standard 19-inch rack. Suggested price is $370.00.

SOUNDCRAFTSMEN
1721 NEWPORT CIRCLE
SANTA ANA, CA 92705
(714) 566-6191

for additional information circle no. 72

NEW UNI-SYNC METERING PANEL

How do you make 24 flashing LEDs an exciting panel? You switch on Uni-Sync's new Professional Metering System, the PMS-1. This peak reading device is capable of measuring line level as well as power output. The PMS-1 is fully adjustable by the rear mounted calibration control up to 200 watts. The 12 LEDs on each channel emit from the center out: green to yellow to red, giving the user instantaneous recognition of equipment status.

The PMS-1 can be connected to mixers, preamps, power amplifiers, or receivers. Just about any place you require an indication of power or level. The inputs are balanced high impedance and utilize 5-way banana binding posts. Of special interest for professional applications: the PMS-1 can be used to monitor balanced lines or power amplifiers with floating grounds or in a bridged mode without affecting the balanced line or causing shorts between two different bridged amplifiers.

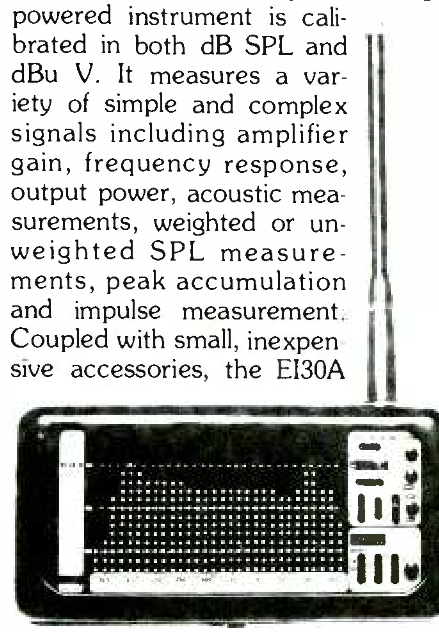
Suggested retail value for the PMS-1 is $149.00, and is available in a black anodized rack-mountable package. Dimensions are 19" wide x $1\frac{3}{4}$" high x $4\frac{1}{2}$" deep. The unit weighs 6 pounds.

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

for additional information circle no. 73

IVIE ELECTRONICS OFFERS PORTABLE ANALYZER/METER

A hand-held audio system combines a real time analyzer (both full-octave and one-third octave) and a precision sound level meter. The portable, high-powered instrument is calibrated in both dB SPL and dBu V. It measures a variety of simple and complex signals including amplifier gain, frequency response, output power, acoustic measurements, weighted or unweighted SPL measurements, peak accumulation and impulse measurement. Coupled with small, inexpensive accessories, the EI30A



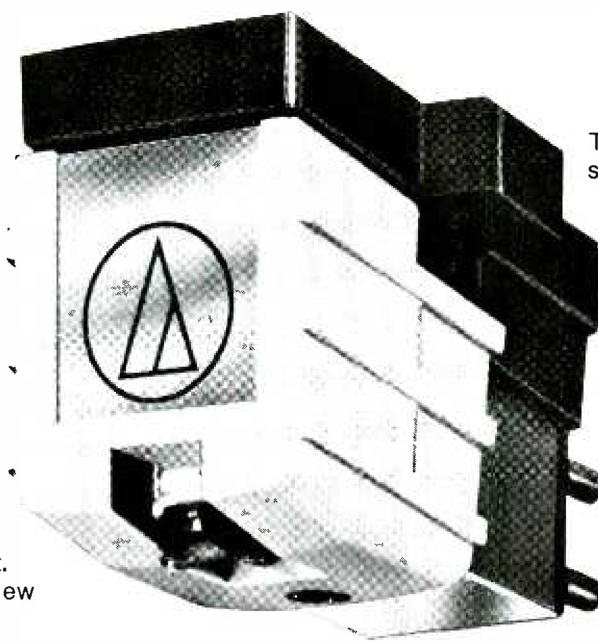
will also provide distortion analysis and reverberation time. Suggested U.S. price is $2,800.00.

IVIE ELECTRONICS
500 WEST 1200 SOUTH
OREM, UT 84057

for additional information circle no. 74

continued overleaf . . .

Audio-Technica rewrites the book on professional phono cartridges.



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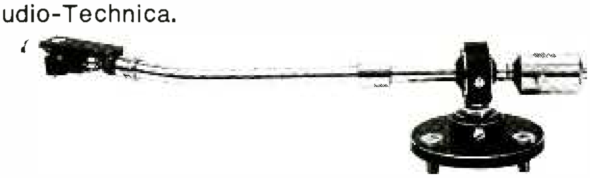
Both ATP cartridges and styli are *uniformly* excellent. When you at last need to replace a stylus, you always get "like new" performance again, and again, and again.

Don't confuse the ATP Series with other "professional" cartridges that are merely modified home units. ATP units don't have to be treated with kid gloves. And yet we haven't sacrificed tracking ability to make them rugged.

The all-new ATP cartridges were specially developed for the working environment. Three models provide a choice of either spherical or elliptical styli. Each cartridge is hand-tuned for optimum performance, with stereo channels matched within 1.5 dB to eliminate balance problems.

All ATP cartridges feature tapered cantilever tubes that combine high strength with minimum moving mass. There's no problem with back cueing, and the brightly colored cantilever tip is readily visible so that you can spot an LP cut quickly and accurately.

ATP cartridges are priced from $25.00 suggested professional net. Write for complete specifications. Try the ATP Professionals on your own turntables. We know you'll be pleased with what you hear. From the thoughtful pros at Audio-Technica.



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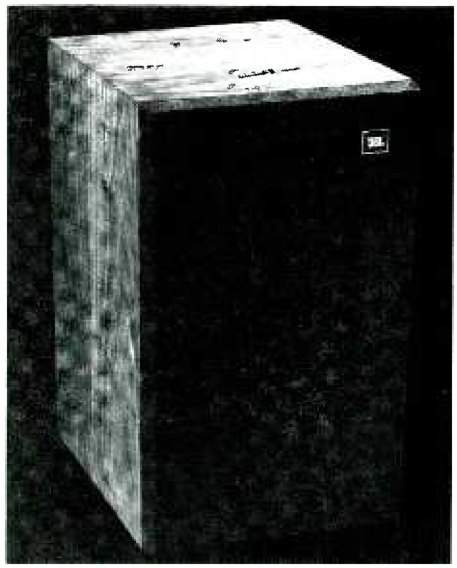
JBL SELF-CONTAINED AMPLIFIER/MONITOR LOUDSPEAKER SYSTEM



A self-contained professional amplifier and broadcast monitor loudspeaker system (the 4301E) has a power amplifier designed to complement the characteristics of the system's transducers and frequency dividing network. As a result, the manufacturer says the output signal which is gained offers highly accurate, efficient and flexible monitoring.

The power ouput capabilities and compact size of the 4301E make it an ideal system not only for broadcast applications, but for mobile recording and editing as well. The system can produce a sound pressure level of 98 dB in a typical 6 x 8 x 10 foot broadcast booth. It can bridge a line-level output from a mixing board or broadcast audio console and thereby substitute for the lower-quality monitor amplifier often provided in such installations.

The 4301E is energized by a built-in amplifier which delivers 10 W continuous sine wave (0.05% total harmonic distortion at full rated power). The amplifier offers completely symmetrical, full-complementary circuitry and wide bandwidth to preserve accurate reproduction of complex musical waveforms. The eight-inch, low-frequency loudspeaker is purpose-built for the system's compact enclosure and features a two-inch diameter copper voice coil and 2.5-pound magnet assembly. Designed for maximum power handling and efficiency, this low frequency driver delivers clean, tight bass under the most severe operating conditions.

High frequency response is provided by a 1.4-inch direct radiator. A 5/8-inch copper voice coil delivers high efficiency and accurate transient reproduction; the small cone diameter and center dome ensure wide dispersion. The frequency dividing network was specially-developed for use in the 4301E and features conjugate circuitry and continuously variable control.

Panels of the enclosure are constructed of 3/4-inch dense compressed wood and all joints are hand-fitted and wood-welded. The veneer is American black walnut, with a hand-rubbed oil finish.

JAMES B. LANSING SOUND
8500 BALBOA BOULEVARD
NORTHRIDGE, CA 91329
(213) 893-8411

for additional information circle no. 78

URSA MAJOR ANNOUNCES DIGITAL REVERBERATION SYSTEM

At a professional net price of $1,995.00, the SST-282 will put digital reverberation and time delay processing within the reach of many users presently relying on combinations of delay units and mechanical reverberation systems.

The Space Station accepts a mono line level input and sends it to a nine input stereo mixer, and to a digital delay and reverberation processor. Eight delay signals from the digital unit are returned to the mixer where they can be combined with the source in any desired combination and ratio. Virtually any pattern of direct sound, early reflections, and reverberation can be obtained by suitable adjustment of the controls. Additional controls adjust parameters of the reverberation synthesizer to permit complete freedom in tailoring the initial delay, decay time, and high and low frequency equalization. The digital processor is switch programmable for many different reverberation and delay effects. The versatile built-in mixer means that the Space Station can serve as a complete reverberation and delay mixdown center without an external console, greatly easing the demand on other studio equipment. The range of reverberation/delay effects extends from the simulation of traditional acoustic spaces, through artificial reverberators, into the realm of special effects. Many delay effects, such as doubling and slap, can be done with new power due to the availability of up to eight delay taps in the mixer.

Recording studios will find the Space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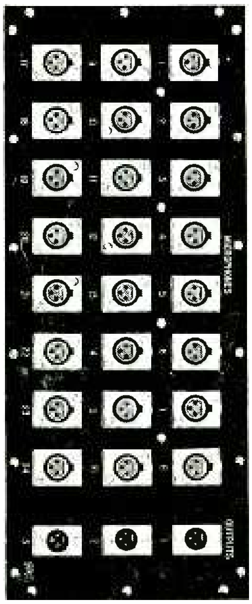

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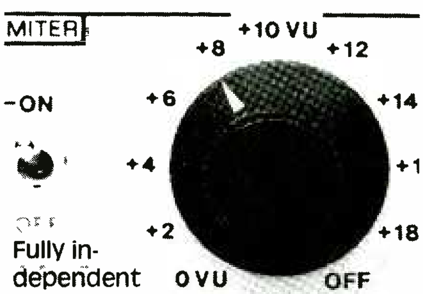

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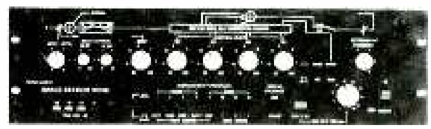
Station optimum for processing dry tracks at mixdown, where delay and reverberation provide a much-needed sweetening and ambience enhancement for vocals and instrumentals. Studios and performing musicians will use it to enhance the sound of their musical signals with delay-based effects from the subtle (doubling) to the outrageous (echoes, long reverberation). Night clubs, discos, and other live performance installations can use the SST-282 to add body and life to sound reinforcement in acoustically flat environments.



Specifications in the delay mode include an 80 dB dynamic range, 0.2% maximum distortion and noise, and 30 - 7 kHz frequency response. Maximum delay time is 256 ms, while reverberation decay time is adjustable from zero to 4 seconds. A peak reading level indicator of 4 LEDs displays signal level in the digital processor so that the input sensitivity can be adjusted for optimum dynamic range.

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The memory will retain such parameters as VCO Tuning, Waveform (sawtooth, triangle, square, pulse), VCF Tuning, VCF Resonance, Filter Keyboard Tracking, Envelopes, Sync, Noise, Cross Modulation, and Volume. This gives the musician 9 (8 program, 1 manual) completely different patches of his own design. The OB-1 is shipped with 8 presets from the factory which include bass, flute, talk box, violin, harmonica, horns, etc.

The performance section boasts a radically new modulation lever, conveniently located for left hand use, with 8 different modulations available at the touch of a switch including pitch bend, growl, LFO, et al. Other performance features include 3 octave transpose, portamento, master VCF, envelope reset, delay vibrato, and red noise modulation. The OB-1's suggested retail price is $1,895.00.

OBERHEIM ELECTRONICS
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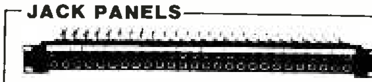

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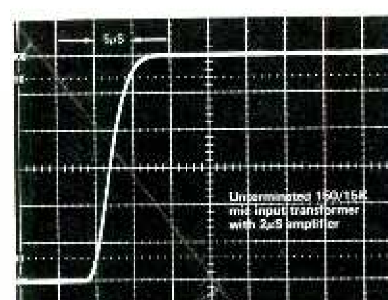

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digital audio report

SOUNDSTREAM

— continued from page 19 —

of inputs that would reveal abnormal noise levels, harmonic distortion, T. I. M., or hum. No prerecording adjustments, such as bias or equalization are necessary. System cabling is carefully planned with close attention to ground loops (which cause hum) and potential noise paths. Once cabled, the entire system, including the recorder, mixing console, microphone preamps, and monitor system, is checked meticulously using the distortion analyzer. Anomalies such as noisy slide pots, high distortion transformers, or noisy microphone preamps are either replaced, repaired, or bypassed. Since the dynamic range of the recorder can accommodate large signal peaks, the use of compression/expansion equipment, which can only compromise signal quality, is typically bypassed.

Presently, material recorded digitally for commercial release is mixed live through an analog console to two tracks of the digital recorder. Most analog mixing consoles do not preserve signal quality as well as the digital recorder. Transformers impose distortion at low frequencies, and in some consoles, at any frequency during a peak level signal. Other distortions have been traced to poorly designed amplifiers. Since each stage of an analog console has a finite dynamic range, internal amplifier stages can saturate while output stages remain within their voltage bounds. The fault here lies with the mixing engineer who sets input channel faders too high yet leaves the master fader low. Current technology allows construction of amplifiers with even less noise than the quantization noise of the digital recorder, although few consoles measured by Soundstream can pass a signal without introducing a higher noise level.

The digital recorder's noise and distortion do not increase above 0 VU as do analog recorders, but the recorder does hard-clip when driven by signals exceeding the maximum input range. Once the gain has been set to avoid clipping, it is not adjusted during a performance. There is no need to increase the signal level during quiet passages. Such gain-riding and/or use of peak-limiters is unnecessary and highly undesirable for audiophile productions. In recordings made using a Soundstream recorder, the engineer has watched the peak-reading LED display on the recorder for peak clipping and has paid little attention to the VU meter on the analog mixing console. The recording session is typically monitored from the recorder's line out which cannot be acoustically distinguished from the feed. (Any confusion can be resolved by following cables!)

Multi-track digital recordings, using standard overdubbing techniques, will soon emerge. Once multi-track material has been recorded digitally it can be mixed through a conventional mixing console, or mixed digitally by combining the numbers as they come off tape. Digital mixing is more precise and allows a two-track master tape to be made without further analog degradation.

The Editing Session

The new methods of editing digital material represents perhaps the largest departure from analog methods. Splicing analog tape yields acceptable results because joining sections of analog tape, which have been cut at an angle, averages the magnetism across the splice boundary, thus reducing transients. However, joining pieces of digital tape can result in peak-level transients. Also, handling digital tape (which puts finger oils on the oxide) makes the numbers on the tape unreadable.

One solution to both problems in an editing system which does not involve cutting tape but rather copies digital data from one place to another. In such a system, the source material is never destroyed, so the splice can be redone as many times as necessary for satisfactory results.

At Soundstream, all editing is done on a small computer system which has been programmed to locate good splice points, copy data, and provide auditioning of completed splices. Both the digital tape recorder and the computer system accommodate a cable which transfers the digital words, as reconstructed from the digital tape, into computer memory. Data in the memory is then stored on removable disk packs (each pack is shaped like an angelfood cake and holds about three minutes worth of stereo data). Once all of the takes that need to be included on the final record have been transferred to disk packs, the computer program which does the actual editing is started. The program accepts editing commands from a keyboard and can be used by a production engineer after only brief instruction. The engineer still must determine where, within two takes of a performance, the musical context will match sufficiently for those takes to be joined.

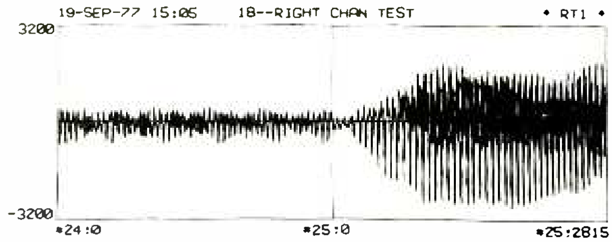
To locate the exact splice points, the production engineer first commands the computer to play the first take and then presses a key on the keyboard when he hears the desired stopping point in that take. In similar fashion he listens to the second take and presses the key again when he hears the desired starting point in the second take. The computer examines the digital data for the two takes in the vicinity of the key strokes and computes the best splice point in each take. The criterion used to determine the splice point is simply that the resulting splice should be as inaudible as possible.

The computer plays the two takes as joined at the computer's proposed splice point. If the engineer approves of the proposed splice, he directs the computer to store the splice information in a table and then proceeds to the next splice. Generally, the proposed splice is acceptable but occasionally the musical transition between the two takes is too abrupt. Then the engineer has several options which include:

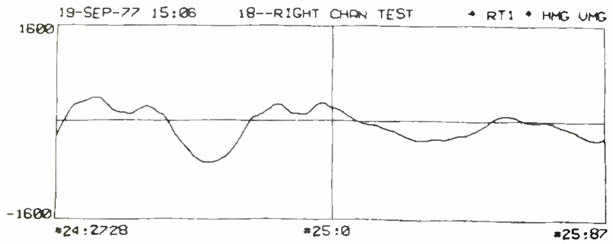
1) Giving new cues to the program and listening to the new proposed splice.

2) Looking at the waveforms of the two takes on a graphics display and visually specifying which sections of waveforms should be joined, or

3) Specifying, in each take, the exact



completed splice at index point 25:0. Top: Left channel. Bottom: Right channel.



Magnified view of figure above demonstrating the continuity in the waveforms across the splice boundaries.

SPLICE TABLE

	LEFT		RIGHT	
SPLICE #1	LS2 (1)	#0;0	RS2 (2)	#0,0
SPLICE #2	LS1 (1) LS2 (1)	#772,1570 #823;184	RS1 (2) RS2 (2)	#772;1565 #823,179
SPLICE #3	LS1 (1) LS2 (1)	#1163,44 #1213,2337	RS1 (2) RS2 (2)	#1163;47 #1213;2340
SPLICE #4	LS1 (1) LS2 (1)	#1493;2700 #1524;424	RS1 (2) RS2 (2)	#1493,2700 #1524;424
SPLICE #5	LS1 (1) LS2 (1)	#1660,580 #1719;1500	RS1 (2) RS2 (2)	#1660;568 #1719;1488
SPLICE #6	LS1 (1) LS2 (1)	#2082;1482 #2147,1597	RS1 (2) RS2 (2)	#2082;1493 #2147;1608
SPLICE #7	LS1 (1) LS2 (1)	#2845;0 #2845,0	RS1 (2) RS2 (2)	#2845;0 #2845;0

A completed splice table generated by the automated editing system.

sample number to be joined (this allows splice resolution to ±10 us).

The Mastering Session

Digital material can be mastered with a conventional disk cutting lathe. Lacquers so produced clearly exhibit greater dynamic range and less distortion than do lacquers mastered from analog recorders. For the mastering session, the digital recorder and master tapes are shipped to a mastering facility and connected to the lathe. During equipment setup, any distortion-causing elements within the lathe, such as input transformers, limiters, and compression circuitry, are bypassed. While mastering, a stereo feed from the digital recorder drives the cutter while the preview channels control groove spacing and depth.

After digital recording, editing and mastering, the final product is of audiophile quality. There are, however, distinct differences from typical audiophile products:

1) Auditioning the digital machine permits equalization and levels to be accurately pre-adjusted, thus minimizing the number of lost lacquers.

2) Because the digital machine provides preview, a record's playing time can be increased without groove echo.

3) The digital recorder may be played at half-speed while the record is cut at half-speed to enhance its fidelity.

4) Lastly, there is no limit to the number of lacquers that can be cut.

Conclusion

When digital audio theory is properly implemented, the digital audio process can produce extremely high quality records. Presently, digital audio is producing audiophile records of the same or better quality as direct-to-disk records, but without the constraints and limitations of the latter method.

Present digital processes represent state-of-the-art technical development, and are more expensive than their analog counterparts, although advancements in digital technology will ultimately allow competitive costs. Technological development in the near future will allow digital data and digital playback equipment to become the standard products for home audio consumers.

WHAT IS AUTO-SET?

AUTO-SET is not a single piece of equipment. Rather it is a system of process control micro-computers designed for the entertainment industry, and manufactured by Harrison Systems.

The first implementation of the AUTO-SET system is the 864 AUTO-SET version 1.0. This version of AUTO-SET is currently available from Harrison Systems and is for use with the Harrison 24 series, 32 series and 32B series consoles.

Additionally, the 864 AUTO-SET V1.0 can be used in any application where control signals must be stored and recalled with data management capabilities. This includes, but is not limited to audio, video, lighting and special effects.

WHAT MAKES AUTO-SET DIFFERENT?

There are four basic differences between AUTO-SET and previous automation "programmers". They are:

Physical Presentation
Data Management
Software Control
Open-ended System

Physical Presentation

AUTO-SET's obvious difference is the physical presentation of the system to the operator. The physical package appears to be a small computer terminal.

Data Management

Data management is the not so obvious difference between AUTO-SET and most previous automation systems.

Data management, in simple terms, is the ability to manipulate the data. This includes the ability to merge or separately use individual components of various data sets.

Data management in the 864 AUTO-SET V1.0 is extensive but is presented in such a way that even a novice operator can beneficially use the system with a few minutes instruction.

The data management capability includes the ability to store up to four independent mixes or dynamic sets of data on one track of an audio recorder.

Data management also includes the ability to store "Snapshot mixes" or static sets of data on a data cartridge machine included in AUTO-SET. Up to 630 individual sets of data can be stored on each cartridge.

NOW AUTO·SET

Software Control

Internally AUTO-SET is a software or more correctly, a firmware driven machine. This means that there are many features and refinements of operation that could not economically be offered with a traditional "hard logic" design.

Open-ended System

AUTO-SET is modular. Future hardware and software modules will be available to perform many new functions.

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