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August 1977
Volume 8 — Number 4

Recording engineer producer

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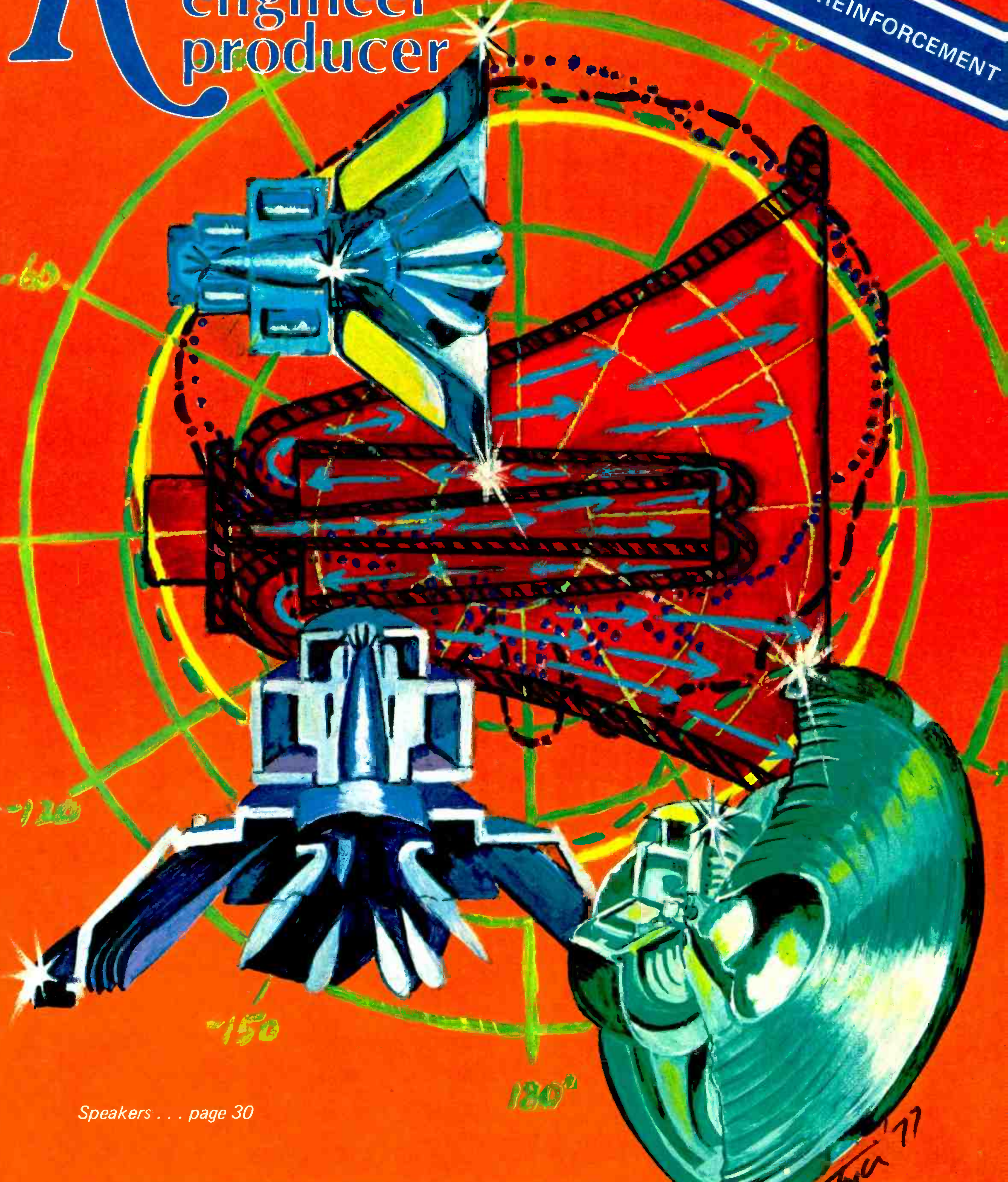

Speakers . . . page 30

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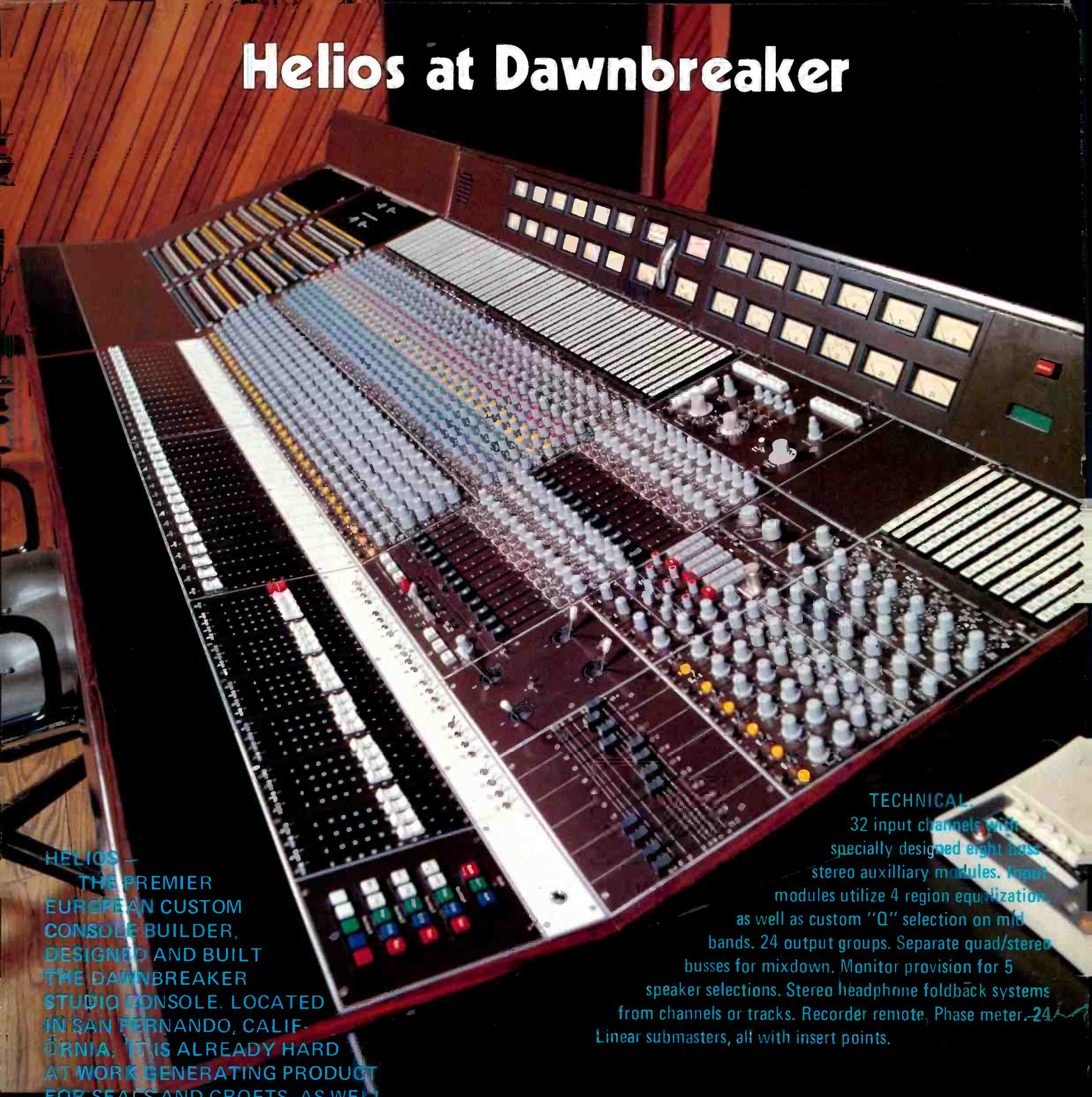

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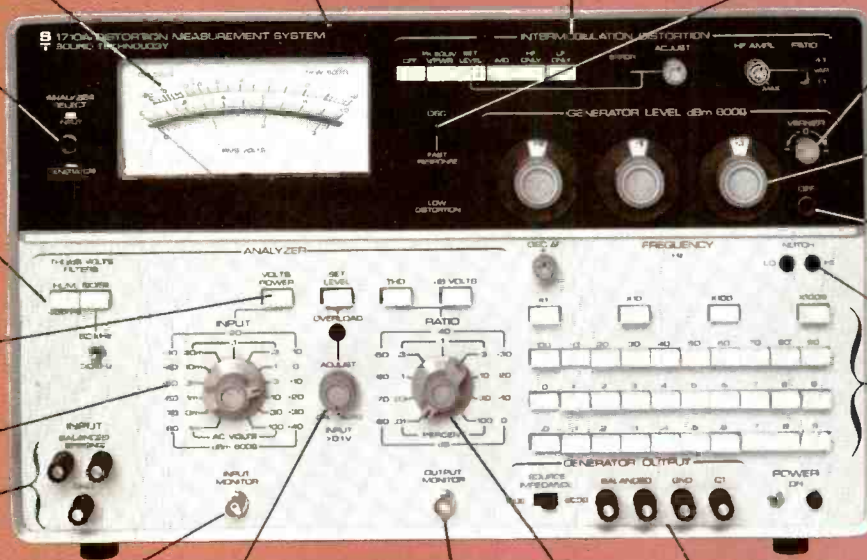

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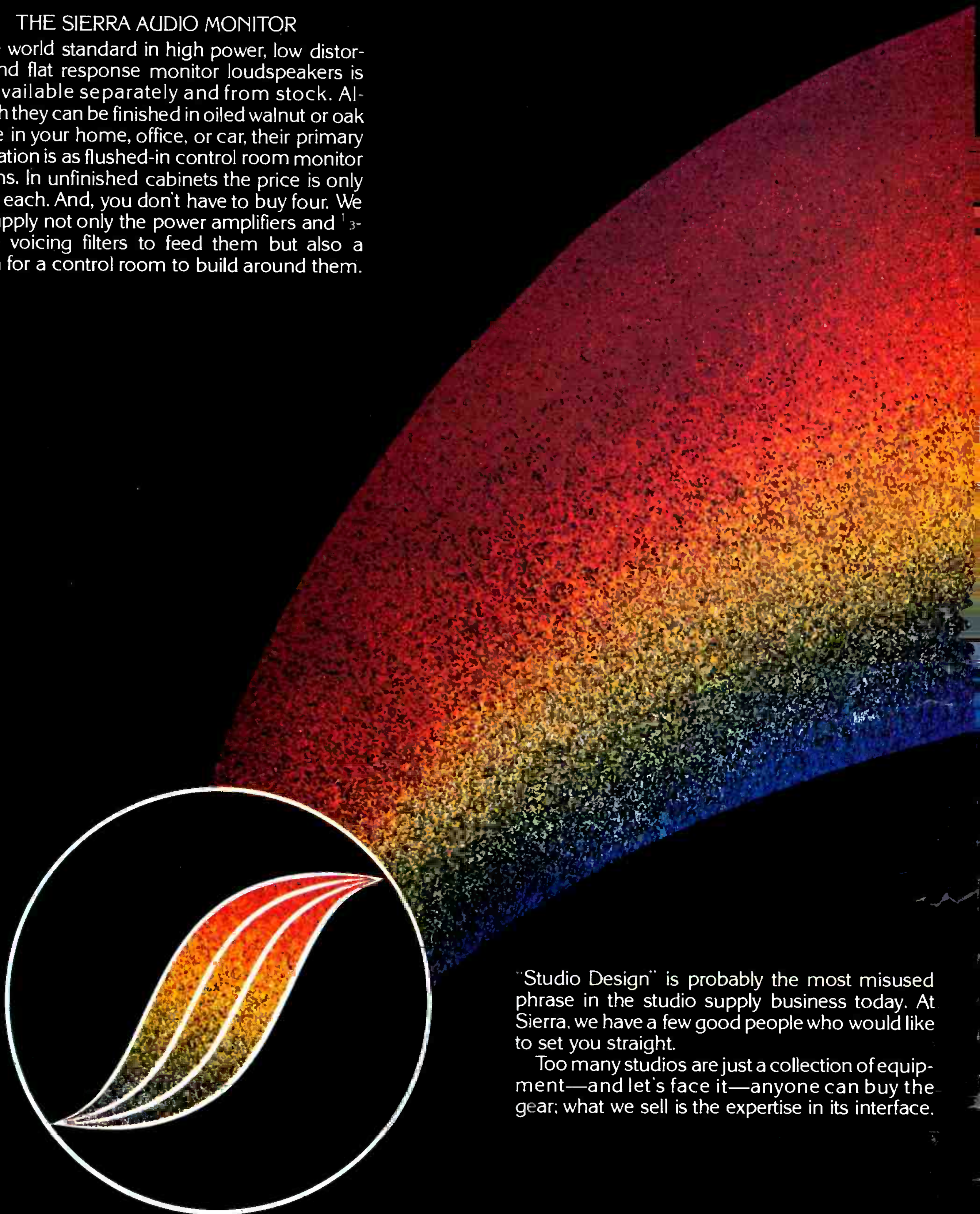

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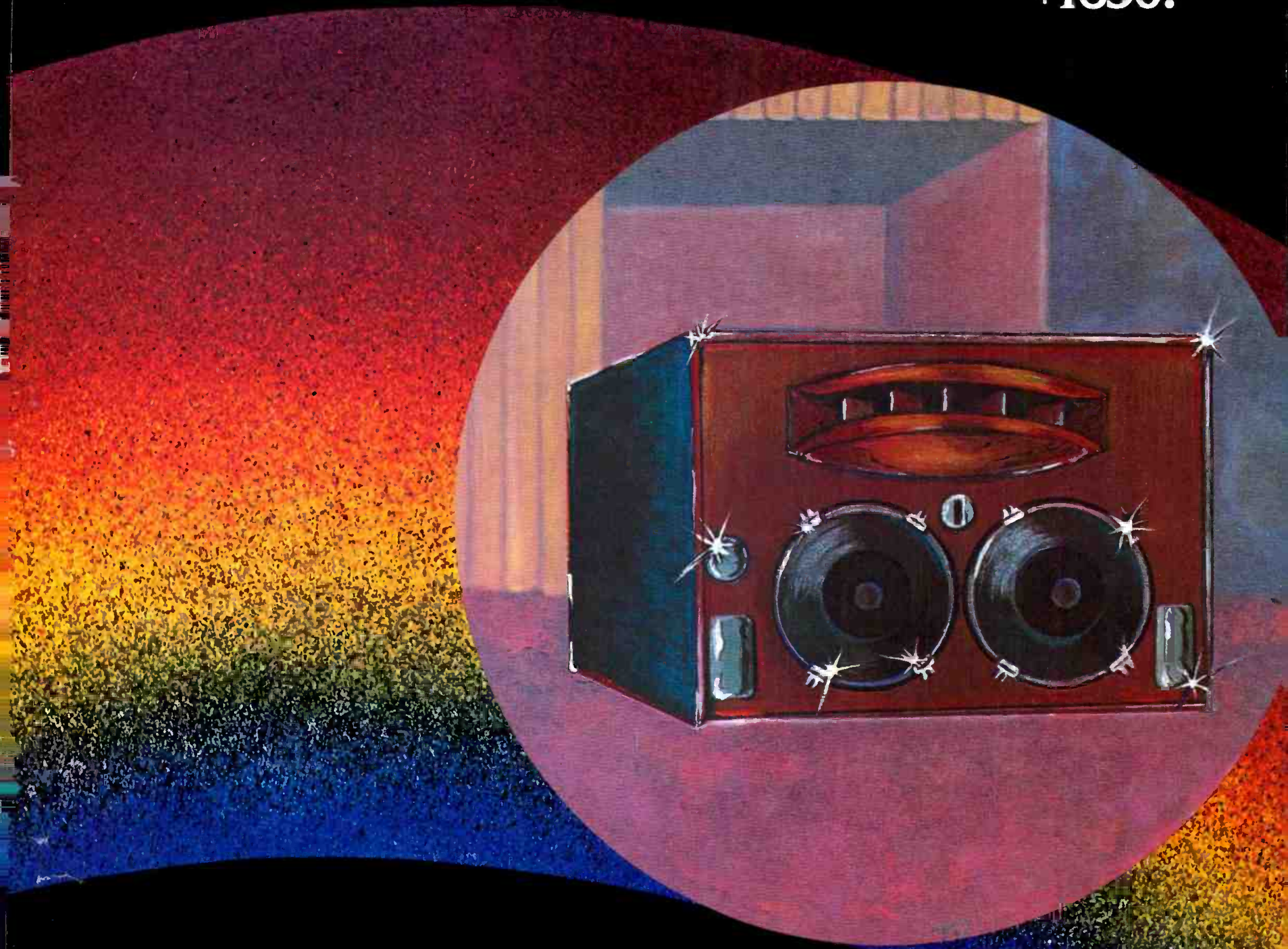

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Sierra Audio

Letters and Late News

from: **WAYNE JONES**
Amber Electro Design, Ltd.
Montreal, Canada

Congratulations on the June '77 issue and its excellent technical articles. Christofer Moore's thorough treatment of digital delay is the most comprehensive treatment on the subject I have ever seen to date.

I was particularly happy to see Alan Fierstein's article "Equalization Myth". I believe it typifies a movement towards a more complete understanding of studio acoustics, and in particular monitor systems.

The advent of the now ubiquitous real time spectrum analyser (RTA) a few years ago gave us a false simplification of monitor system "tuning" or equalization. Armed with a portable RTA and an equalizer filter set just about anyone could "tune a room". The lack of subjective acceptance of such rooms resulted from their wide variation in true performance, although they might have appeared "flat" on a RTA. This spurred the development of additional work in both acoustics and electronic portons of the monitor system. Active traps, geometric designs, improved enclosure mounting and other refinements evolved out of the acoustic research.

Improved and lower cost instrumentation has furthered the development of both the electronic and acoustic properties of the monitor system. New spectrum analysers provide higher resolution in the frequency domain, reverb time measurements can be made in a variety of ways using recently introduced test equipment and phase measurements of both the electronic portions and the speaker system itself are now relatively easy.

The RTA presented a simple, easy to interpret picture but we now see it hides a variety of system faults even more serious than the amplitude response variations it does show. Unbalanced reverb, poor phase response, steep, narrow amplitude variations can all have a severe effect on the "sound" of a monitor system yet they may not even show up on a RTA. Using a swept frequency analyser and other types of XY plotting equipment the true performance becomes more apparent.

Plots of amplitude vs. frequency, phase vs. frequency and aplitude vs. time take longer than the simple RTA presentation. They also require interpretation or perhaps it should be called "Applied Engineering". The final audible results of the additional effort however, most dramatically demonstrate that all aspects of a total system must be measured if we are to approach the state of the art in acoustic performance.

---

from: **JAMES B. SHARPE**
Chief Engineer
Movie Tech
Hollywood, California

Thank you for your excellent article. (*"Post Production Audio Sweetening", by Paul Sharp, June 1977 R-e/p.*) I shall utilize it in orientation of new personnel and uninformed clients.

I do have a bone to pick with you. The utilization of S.I. Units and their abbreviations may seem out of place in our vernacular but there is no good reason not to incorporate them when it removes ambiguity from our communication.

You and I know what you mean by ". . . a sync pulse, usually 60 cycles . . .", but does the displaced aerospace technician? Let's examine that statement closely:

Sync: An abbreviation for synchronization. A locking or timing function, a way of bringing two independent systems together in time. (Rather nebulous.)

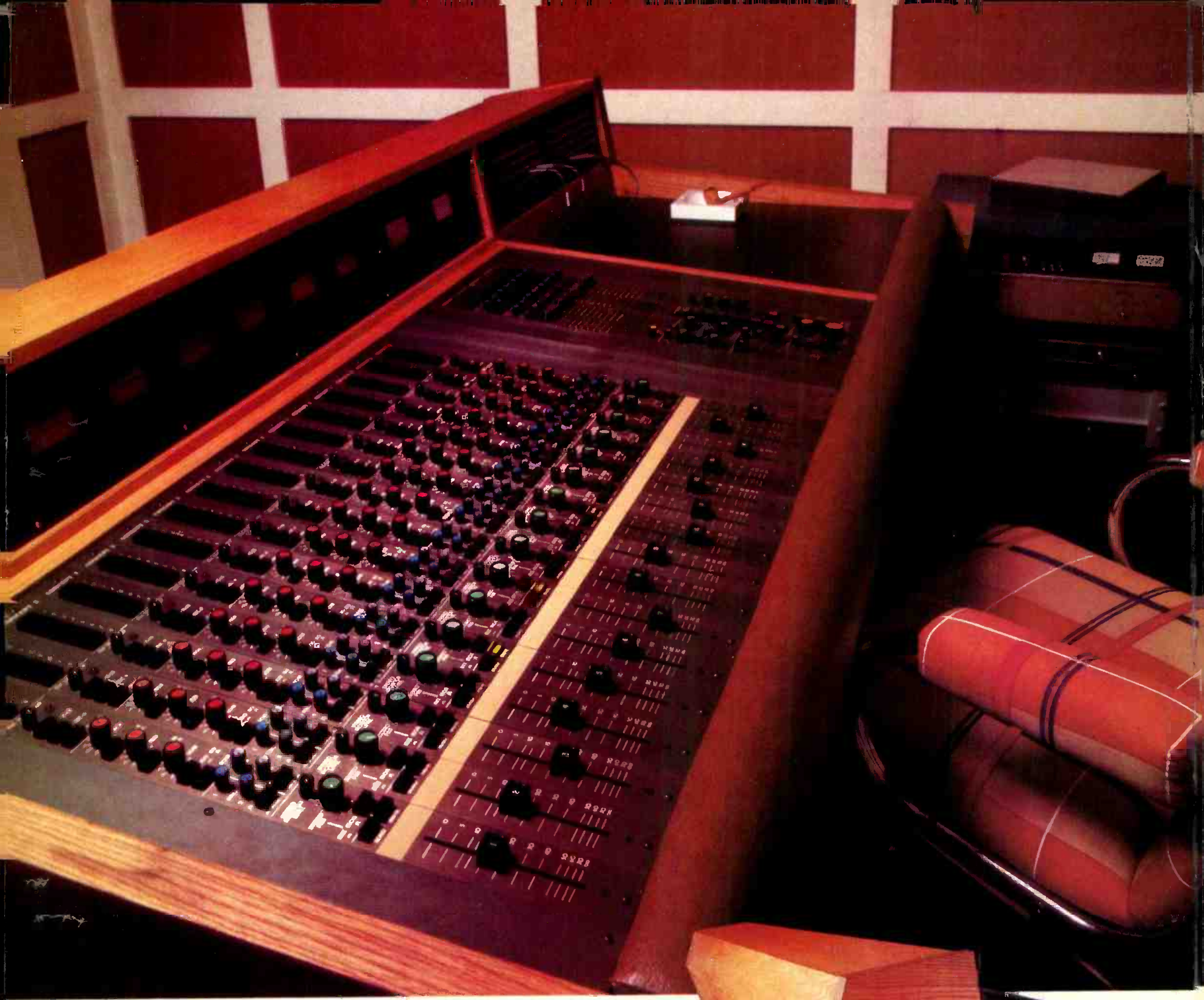
Pulse: A change in amplitude, with undefined shape, direction, duration, or duty cycle. (Obscure.)

(Our technician is now dealing with an idea that is both nebulous and obscure, a good definition of a fog.)

60 Cycles: Our technician should instantly have made the correlation between this and the wall duplex plug. The correlation is tested against the content of text with no confirmation. The only conclusion possible is there may or may not exsit a correlation so the key may rest in the word pulse, or sync.

(We have led our technician down the garden path to wonderland.)

A quick scan of the memory bank reveals a sync pulse system used in video and a text correlation! With three sync pulses, color burst, horizontal, and vertical to select from, the obvious selection is the vertical interval sync of 59.94 cps. It's not cycles though, close but not close enough for sync. Don't we know it!



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None of this is the idea you were trying to communicate. You may insist that the word cycles does. I don't think so. Since I have the book open let's return to video and the color burst. It is 8 cycles! Not 8 Hz., but 3.579545 mHz., or a burst of 8 cycles at 3.579545 mHz. There is a fundamental difference between cycles and Hertz. We should observe this difference not ignore it or we are faced with meaning what we say, not saying what we mean. To wit, 60 Hz. sync pulse. Preferably a sine wave of continuous duration that may be obtained from the wall duplex plug if the other system you are trying to sync to is synchronous to the wall outlet also.

I really did find your article very good and intend to utilize it, even if you do spell your name wrong!

Reply from: **PAUL SHARP**

You may have my half of the bone.

60 Cycles is the standard sync pulse reference in film. It is used with mag tape, which has no sprocket holes. Videotape, as pointed out in the article, uses different reference frequencies for synchronization.

A subject I neglected to mention was motor systems used in film sound with sprocketed magnetic film (and the picture), where a large distributor drives any number of motors, controlling the mag film transports of reproducers and recorders (and the projector). The subject could become involved, but suffice it to say that "locking" all motors onto the same distributor system ensures synchronization while work proceeds.

I would like to take this opportunity to add some corrections. The title of the article and the labels for the diagrams were supplied by someone other than myself, and they are not in every case accurate.

The label under Figure 2 is particulary strange. A correct label would be "Possible dialogue units for a film production."

One typographical error needs attention as well. On page 41, under the heading "Lay-over", the first sentence should say, "The final edited master from 'on-line' is copied back to a smaller videotape format, usually 1" helical scan . . .", rather than one-half inch.

Most film sound personnel are familiar with the film sound post production process, and many Hollywood videotape people are familiar with the type of audio sweetening described. My hope with this article was to shed a little light for each regarding the "other medium".

WESTLAKE HONORED FOR $2 MILLION OF 3M-MINCOM MULTITRACK RECORDER SALES

Pictured is Glenn Phoenix, Westlake president, accepting plaque from 3M-Mincom marketing manager, Bob Brown commemorating the company's second million dollars of sales of 3M multitrack recorders.

STUDER TAKES OVER REVOX DISTRIBUTION IN USA

Willi Studer's takeover of USA distribution of Revox brand tape recorders as well as other Revox audio products was announced recently by Raymond Updike, vice president and general manager for Studer in Nashville.

The name of the domestic operation has been changed from Willi Studer America to Studer Revox America, Inc. to reflect the addition of the Revox products to the company's direct responsibility.

Designed to provide Revox dealers and owners the same service presently enjoyed by Studer professional users, the changes include the setting up of regional service centers at 155 Avenue of the Americas in New York City, and at 14046 Burbank Boulevard, Van Nuys, California. Warehousing and major service on Revox will be handled in Nashville, where the firm's facilities at 1819 Broadway have recently been doubled in size to accommodate the increased volume.

"Domestic sales of our professional tape recorder products have increased dramatically since we went direct on the Studer line in 1974," says Ray Updike, "and we expect to achieve a similar dramatic increase in market share for our Revox line in a short time. We're going to keep essentially the same rep orginization, but with our professionally oriented philosophy and dealer support on sales and service, we feel that they'll be able to do a better job with the Revox products than was possible previously."

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Audiotechniques' president, Hamilton Brosious, has announced the formation of Audio Marketing, Ltd. The new company, a wholly owned subsidiary of Audiotechniques, Inc., has been set up to supply innovative products to the professional recording industry and to the growing "Sound Reinforcement" and "Semi-pro Audio" markets.

Heading up the new company is Richard Anderson, former Product Manager for Audiotechniques. In the new post of General Manager, Mr. Anderson will be responsbile for establishing a dealer network in addition to developing or discovering new audio products.

Products and manufacturers already represented by Audio Marketing, Ltd., include: Allen & Heath mixers; Brenell tape recorders; H/H amplifiers; Mastering Lab studio monitoring equipment, including Big Reds and Super Reds; EMS Vocoder; and other products. Audio Marketing, Ltd., has exclusive U.S. distribution of these products and there are plans for selective addition of other new products.

Audio Marketing, Ltd., will soon be moving into expanded facilities in the Audiotechniques' building in Stamford, Connecticut.

Literature is available on all of the products handled by Audio Marketing, Ltd., and dealer inquiries are invited. Wrtie or call:

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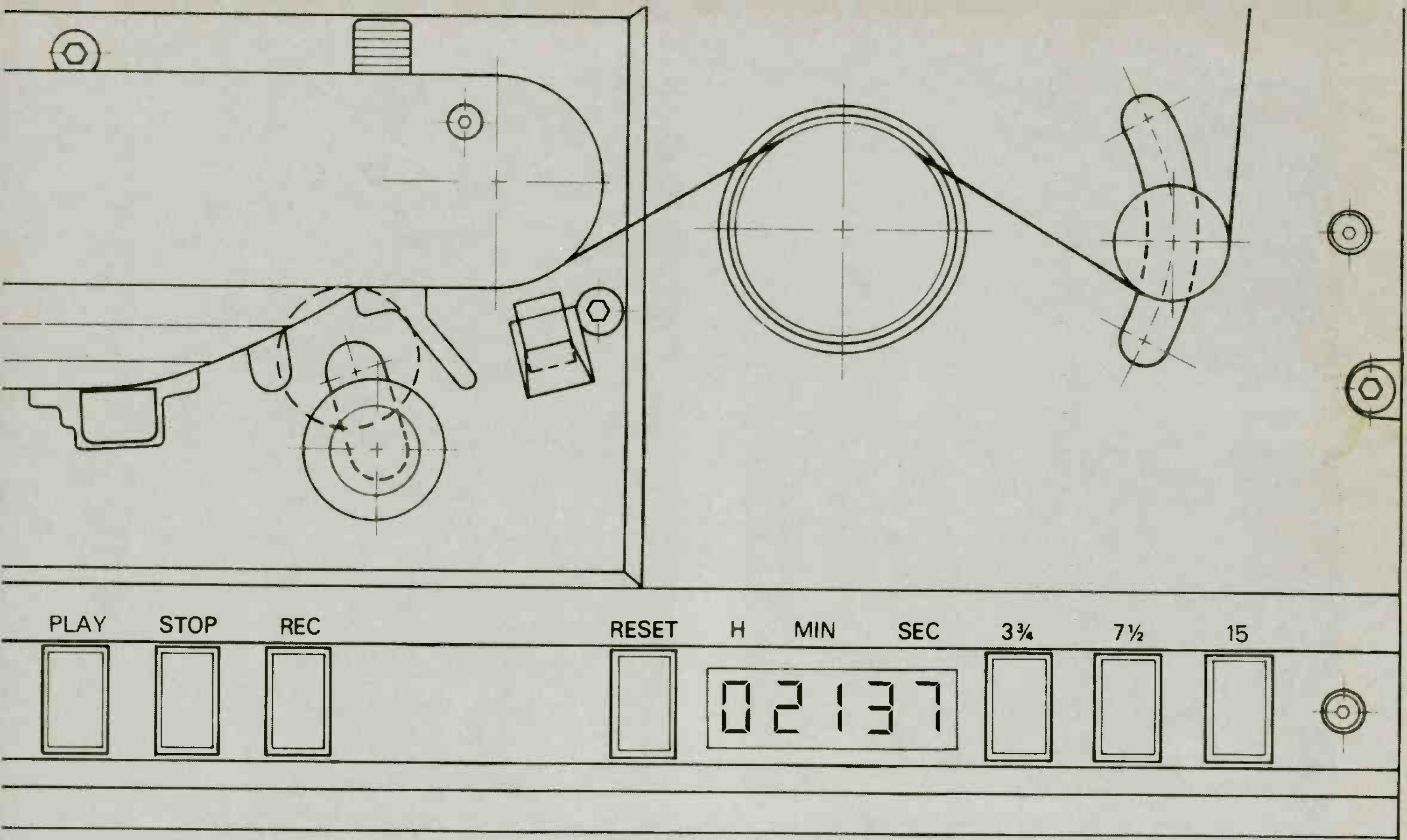

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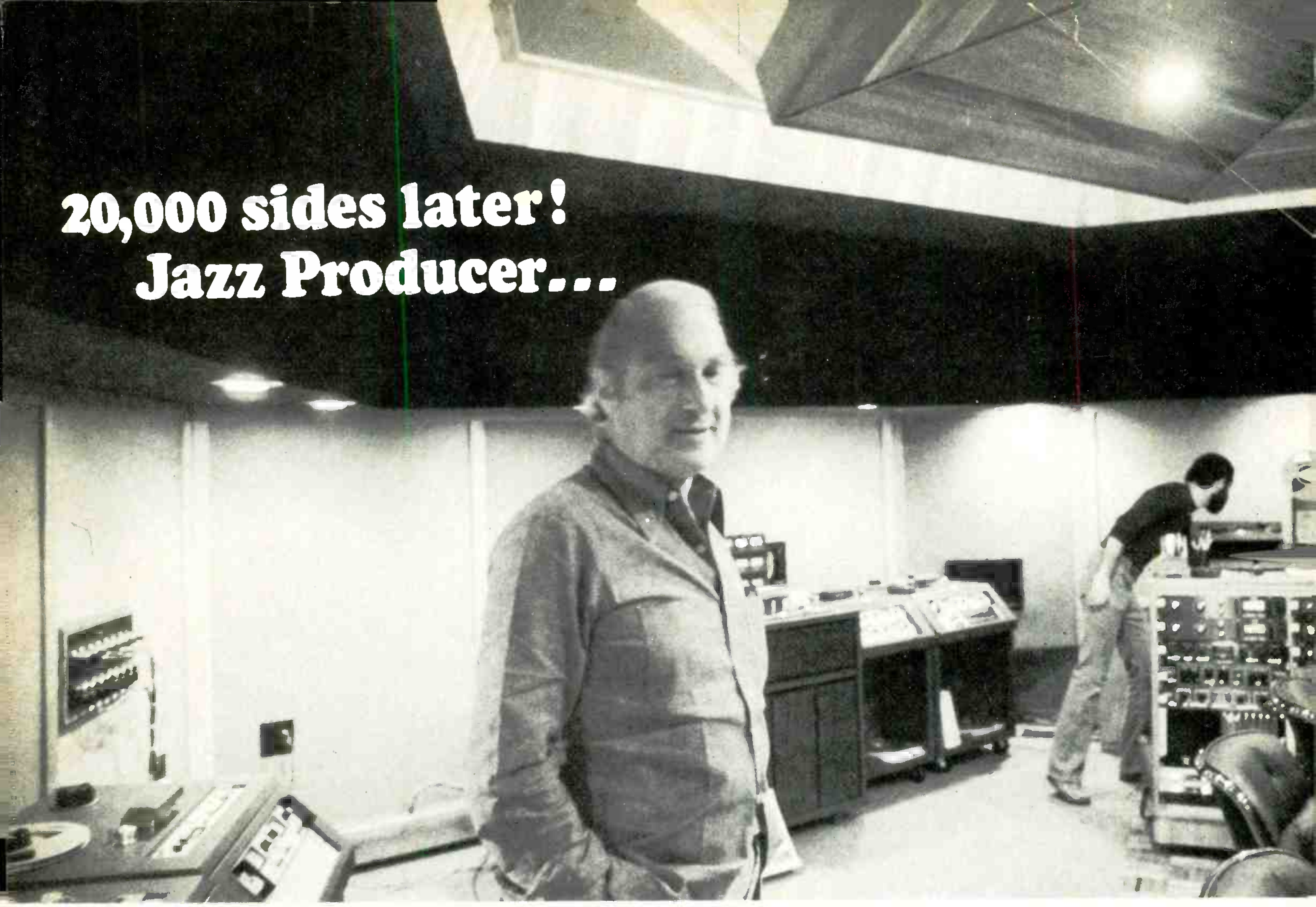

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NORMAN GRANZ

by HOWARD CUMMINGS

Norman Granz: I was born in Los Angeles and went to UCLA, liked jazz as a student, and became friends with some jazz musicians. I primarily became interested in the sociological side of things and the fact that there was a tremendous amount of segregation and prejudice in L.A. at that time (late 30's – early 40's) and I had the idea to combat racialism through the use of jam sessions. Instinctively the type of music I liked, other people liked. And I felt the musicians ought to have a place to do what they did best – just to sit down and jam.

Howard Cummings: Were you influenced by Art Tatum, Charlie Christian, Louis Armstrong, etc?

Norman Granz: I listened to *all* jazz then. It was just the whole *idea* of jazz and it was just impossible to separate jazz from black people. Most of the people I was friends with, mixed with, and worked with were black musicians, with some exceptions. I was very good friends with people like Nat Cole, Lee Young (Lester's brother), etc.

After the war, I began doing jam sessions here seriously in night clubs and again I had rules that you must mix audiences (racially) because, strangely enough, the night clubs on Hollywood Boulevard were just as rigid as those in Mississippi.

I built up a whole circuit. Monday night would be the 331 Club, Tuesday would be another club that was usually closed, and it was good for the employment of these musicians, too. Dexter Gordon used to play a lot of them. Then I gave my first jazz concert at the Philharmonic Auditorium in '44. It worked well enough in L.A. that I decided to take it to San Diego and San Francisco and before long, I was touring with it. I recorded it, of course.

Howard Cummings: Did you have a special crew you worked with?

Norman Granz: No, I couldn't afford it and I would only record concerts in concert halls, not clubs. The first things I recorded were not mine. The Armed Forces Network with Jimmy Lyons, who now runs the Monterey Jazz Festival, recorded the Philharmonic for AFN. I got a copy of it and it seemed a good idea to put it out with all the noise, mistakes, and applause – it was a documentary. I took it to a lot of record companies in New York and they thought it was stupid to put something out with noise. Finally one small label, Ash Records, decided to put out JAZZ AT THE PHILHARMONIC – VOLUME 1, which became rather famous and sold very well. Then I began to put out other volumes and recording other remotes. I tried as much as I could never to tell the musicians they were being recorded, so I began recording 3 or 4 cities on the tour in hopes of getting a more natural kind of play. That worked so I could pick a particular city that did best.

When the big companies decided it was a good idea to start remotes, they faked applause and they added things to it. Even the companies that bought my masters would cut out solos. I never did that. The concert came out just as it was done – in sequence – and I concede it may have been more commercial to re-arrange things but I felt if you were going to do it, you should do it honestly. Sometimes I put out recordings that may not have been very good technically, but I felt the music was so good then. And I never added applause. If there were only 10 people there, that's the way it came out. It will be the same way with my new label, "Pablo Live". There's no faking.

". . . separation . . . that's fine if the form is the important thing . . .
I consider content to be of total primacy . . . I understand leakage."

Howard Cummings: When you would put an album together, would you inter-cut between cities?

Norman Granz: No, because we were still on disc. The idea occurred to me many times because sometimes we would never get *enough* in a city. In those days of 78's, putting out three of them would only give you twelve minutes of music with four minutes per side.

By 1957, the whole thing had become repetitive and I was using much the same people and I felt, "Who's interested in more than 15 volumes of JAZZ AT THE PHILHARMONIC?"

HC: What do you feel is important in presenting jazz?

NG: I've never liked the mass approach to jazz like the Newport Jazz Festival. I feel that's more quantative. I think I've always tried to keep my sessions as I think musicians would like it. I could never put on a jam session at Radio City Music Hall with 30 musicians on stage because nobody gets a chance to blow, and nothing is knit tightly. I always try to do what musicians *naturally* would do. This meant of course that I always would consult with the musicians involved. For example, I did some things at Montreux this summer with two jam sessions, and I spent a lot of time considering just the addition of a fifth horn in the session. I talked to all the people in it asking "What do you think? Should we add a trombone or not? What do you think about this trombone player? Will it really work?" One man, in my opinion can throw the whole thing out of kilter — the same with one too few. I have to think of the competitive attitudes I like among my musicians and how one's going to spark on the other. The whole reference point is: Is this how the musicians themselves would want to do it?

HC: So there is a pre-production meeting of sorts.

NG: In the sense that I'd been writing up all kinds of combinations and I'd call up Oscar Peterson and Basie and ask them what they think — it's give-and-take.

When I record a concert, I never think of the recording aspects — ever. I think that's a fallacy. You have to do a concert primarily for the public that's there, you're not doing a record session with people invited. I think that's a big mistake. I know many of the rock acts will do a live performance, that's essentially a recording date — and the public's invited. That's turning it around in my opinion. I'm doing a show designed to please the public and the musicians are chosen musically. The fact that I'm recording it is the reason why there might be a high rejection rate. What comes off may please the public, but may not be good for record.

Ella's (Fitzgerald) a good example — she's difficult to record at a concert because she's forced to do a lot of tunes that people want to hear her do that she's already recorded. I could say, "Ella, I want you to do twelve new songs on this concert because I'm recording it". But I'd screw up the whole concert. People want to hear MACK or something like that. So I don't do record sessions with the public, I do concerts which are recorded — and there's a big difference.

HC: How about your "concert" set-up?

NG: Sometimes with the set-up, my recording engineer will say to me, "I can't get that separation." And God forbid if you're televising it — you've got another hassle. I construct my set-up as a concert set-up. What it is is: Where is the musician going to hear his other people best? When (Oscar) Peterson plays with Joe Pass, he wants Joe behind him and the bass player next to him so they can hear each other. But this is a terrible set-up for recording.

HC: How does recording your jazz differ from pop or contemporary?

NG: In pop, the sessions I've seen, the technique is very, very important. The sound is very, very important and everything, to the best of my knowledge, is layered. I don't know if there are any live sessions done.

For example, when I recorded Ella Fitzgerald while she was under contract to Warner Brothers, I had secretaries call me up asking if they could come to the date and I said 'no' because it's a live date. They'd never seen a live date in their lives. All they know is singers who come in with earphones . . .

HC: on Tuesday . . .

NG: and played back. I never record Ella that way. Musicians are there and it's done right off — that's it.

One of the things I do too that's I guess rather unique and different from other jazz sessions: I do an LP in a day. It's rare that I extend over a day. So most of my sessions take 3 - 4 hours and that's a whole LP. Exceptions could be a Basie big band. It's rare that I take more than a day and the musicians are delighted. I pay the men for nine hours work for being there only three hours.

You're allowed, in a three hour session, 15 minutes of music. So in effect I could tell the men they've got to give me nine hours of time to get 45 minutes of music. Well, if a man can give me 45 minutes of music in 45 minutes, he gets paid for nine hours work — he's happy and I'm happy. They love my sessions.

HC: Do you feel you're in the same category as Creed Taylor or Manfred Eicher? (Jazz producers)

NG: I think from what I understand, is he (Manfred) places great emphasis on the brilliance of his records, the recording techniques. He pays a great deal of attention to that. Maybe my stuff by comparison is, if you like, "sloppily" done. But then what is casual or off-the-cuff is no less valuable or important. I think it's the "truer" way to record jazz.

I think Eicher records the type of people who require this repetitive attention. I record jam-sesssion people. He records people who have probably worked out their things very, very carefully and profoundly and I think the re-doing of a Keith Jarrett's works are important for Keith because it's almost quasi-classical whereas Oscar Peterson, Joe Turner, or Dizzy Gillespie go in and play. That's my concept of jazz. I'm consistent with that concept — with the *people* I record and the *way* I record.

HC: Is there anything you look for in a studio?

NG: Not particularly.

HC: Is there such a thing as a "jazz studio" for you?

NG: To my way of thinking — no. I'm not saying they don't exist. I don't know of any. All I care about is I like a lot of room. I hate small rooms. But I don't care where I record.

The very first studios I used and liked the best were Radio Recorders (L.A.). Their studio B was always the best studio in the 40's with (Art) Tatum. Then most of their mixers moved over to Capitol when they built the Tower and I did a lot of my big sessions there — most of my Louis Armstrong.

HC: Do you like the studio in Montreux — the Anita Kerr studio?

NG: I don't use any studio there. I've never recorded in Montreux, I just do concerts there.

HC: How is Montreux set-up for concerts?

NG: There's a big room in their casino which holds a couple thousand people.

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". . . excluding classical, or my type of jazz, [there] is a sameness in the sound of a record —"

They have a platform stage. The engineering portion is in another part of the building and they're all televised for the public. I usually stay downstairs in the musician's room and watch on television.

HC: What about engineer qualifications?
NG: I look for one that will keep his hands away from the knobs.

HC: Just do an immediate set-up and let it ride?
NG: I expect him to do something if they over-cut — if it's too loud. But essentially I like an engineer who allows the musicians to use their own dynamics. Now admittedly today, we cut both ways and I take advantage of the fact that I cut 2-track at the same time I do multi-track so if anything is lost, it can be brought (mixed) up. And I'm very careful in what is brought up, in the sense that I don't want the engineer to inject what *he* thinks is the way it ought to be mixed. I try, as much as I can, to consider how the artists want themselves to be heard.

As an example, I have a particular attitude on how I want my singers to be recorded. Most engineers tell me that's not the way they do it. Most records have a meld where the voice is mixed with the arrangement. If you listen to a lot of the pop things and especially rock, I can hardly hear the lyric — the thing is all submerged.

HC: Some are noted for that.
NG: Yeah. I like my voices *way out* in front, way, way out in front and I want Joe Turner to be heard way out in front. Well, you do as much as you can with a set-up to insure that, but sometimes you still can't get enough of it in front. Sometimes on the mix I may want to help there. And I don't isolate him — he's right in the middle of the whole rhythm section.

HC: In a circular placement?
NG: It depends. He might be facing the piano, he might be next to the piano. I use Joe as an example because he's so traditional sometimes he comes in with just the rhythm section and no horns and then it's easy. There's just the guitar to worry about. When the horns are there, he has to face them because everyone wants to hear the rhythm section and everyone can't get next to the piano.

But there's a minimum when there's any mix necessary. It's pretty much done at the session.

HC: Again, regarding engineers . . .
NG: I have engineers who are very sincere wanting to do a good job from *their* point of view and that's why I work with very few people. In the recent past when I sat down to record with some young engineers and would ask for more bass, they'd say, "Don't worry about it. It's on the tape". Now I record primarily with RCA in New York and they have an engineer there, Bob Simpson, who I'm comfortable with.

When I did the MACHITO (Dizzy Gillespie-Chico Farrell) album, they (the musicians) chose a studio called Generation Sound (NYC). The room was very tiny and they crowded about 25 people in — it was insane. Dizzy was literally in another room. I felt terribly uncomfortable there. Theoretically I was the producer and I worked with a young engineer who had very fixed ideas about how he wanted to do it and I said to Dizzy after the first night, "I really don't feel comfortable about this and I doubt that it's going to work because everything's crowded and it just doesn't feel right to me." And he said, "If you like, we'll go over to RCA." So he called me back about two in the morning and said, "Listen, I've talked to Chico and Chico is really more comfortable at that studio." (Generation) I said, "Well, fine, you do it there," and I didn't show up for the other two days 'cause I had nothing to say. But I didn't intrude on the artist. That's where he wanted it, that's where he got it.

I really don't like the primary position engineers have been put into, not by their own choosing, but by the kind of production going on today. I read about producers who spend six months doing an album. I can't even understand that. I produce anywhere from 40 to 50 LP's per year. Maybe they don't sell a lot, but I'd be willing to put their musical values up against anything that sells in the millions. Certainly I could demonstrate it statistically by the longevity of my albums or other jazz albums. But I can't envisage that any 40-45 minutes of music has to take six months to mix. I can't believe it.

I think the trend of the record business today, especially in pop or non-jazz, has so much emphasis on the technical side of recording that I think the content is lost for the form. I know when I used to record in the late 30's on 16" disc, the contribution of the mixer took place on the session and as a producer you would say, "Give me more bass, less piano," or whatever and because everything was mono, what really happened was the genuine master.

Before that, the real differences arose with the set-up. I understand perfectly the ideal situation for an engineer to try to get perfect separation and that's fine if the form is the important thing. But since I consider content to be of total primacy, I always have problems, especially with new engineers, when I have to break up the entire set-up and re-arrange it. In jazz, at least from my point of view, the interplay among musicians is very important — especially the rhythm section because they have to hear each other. I understand leakage and the fact that the drummer might leak into your piano. But isolating them or putting a singer into a booth for example, you achieve that separation and purity for the mix, but I think you lose a certain ingredient that makes for the essential feeling for jazz. If I do a blues singer like Joe Turner, I record Joe as nearly as I can to the way I recorded him 30 years ago.

That's not to say that there hasn't been progress in recording. I'm not sure that it's terribly progressive *musically* going from mono to stereo. I don't see that stereo has in any way made jazz any better. I personally don't care for stereo, because when I go to hear a concert, I hear it mono. I don't think anyone even sitting dead-center in a hall hears any big stereo.

HC: A few years ago there were "back to mono" buttons.
NG: I guess, for all purposes, what I record so de-emphasizes stereo that you could call it "modern mono".

HC: Do you consciously avoid stereo?
NG: I don't feel one way or the other about stereo. I don't go looking for it — I don't care for it.

There's another profound point I ought to make and that is jazz musicians have a wonderful sense of dynamics and especially a sense of dynamics among each other. In Duke Ellington's big band for example, his trumpets had their own thing going as opposed to the trombones, as opposed to the reeds, as opposed to the rhythm section and the way they all meshed together is the way they heard each other. The lead trumpet player heard the other trumpets, and in effect, what you heard was as if you were at a live performance. But the idea of an engineer bringing up the third trumpet or second trombone leaves you to the mercy of the engineer's sense of dynamics — or even the arranger's.

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"These groups, and I'm not knocking their commercial success have become involved in a tremendous conformity –"

kinds of dates – a Red Seal date for Victor and an Illinois Jacket (sax player) if you like. I think a skilled engineer could put himself in a position of what he was doing more than making what he was doing go to *his* point of view. I think the result, in the broad mass of records, excluding classical or my type of jazz, is a "sameness" in the sound of the record.

I've heard some record companies enjoy that – that they have a special identification. But I wonder what that does to the artist in his own sense of identification and development. Today you have groups, but nobody knows the people in those groups. It's very strange. Everyone knew Coleman Hawkins. The fact that he played in Fletcher Henderson's band in no way obliterated the fact that there was a great sound in the band. But everyone knew that Benny Carter was in the band as well. Everyone knew that other people were in the band. These voices were all individual voices. But today, unless you really know the market, I wonder who is in Led Zeppelin and The Eagles. The Beatles were an exception – but I don't even know who's in the Rolling Stones. The point is: These groups, and I'm not knocking their commercial success, have become involved in a tremendous conformity. That makes for marvelous mass sales. And the kids who thought they were being different from the older generation, in its own way, are even more the same than older people. There was more unanimity of point of view at Woodstock than one would find in older people. I think that pervades music so that the people who make the money, and they're not kids by the way, want this and want the parts interchangeable.

HC: How do you feel about multiple takes?

NG: That's something I'm dead against. I try to get it on the first or second. I'll defer to an artist and most of them wouldn't dream of doing a lot of takes on something. Once in a while you might get someone that really feels what he's played didn't go down very well. In my kind of jazz, where the emphasis is on an improvisational quality, there's very few dates I do where things are "read".

I think when an artist does a lot of different takes on the same number, he doesn't do it better, he does it *differently*. With Charlie Parker or Lester Young, if I told them to play the same thing again, they couldn't do it, even if you asked them to do it. It would simply be a different version. Now they may like one version over another, but I can't say you *improve* something. When I did PORGY AND BESS with Cleo (Laine) and Ray Charles, everything was *written out* so if it had to be done over, they did it essentially the same way but it was technically better. That's different than someone coming in without music and just improvising.

HC: If you and the musicians are not getting what you want in the first few takes, do you put it aside for another day or another hour?

NG: Sure. It's been my experience where musicians will put it away, and if it's something he or I felt was *pretty* good but we didn't agree on, we'd come back

"I don't think George Benson, who's a very good guitar player, is playing jazz when he's doing "Breezin'" as a single, even though he's jazz rooted –"

to it and our attitude might change for the better.

HC: Once you have something on multi-track, do you like to mix right away or do you like to let it slide for a week or a month?

NG: If I had my choice, I'd like to do it the very next day or the same night. I like it when it's fresh in my own mind. I take notes during the session and many times I'll ask the engineer if something was there that I wanted to hear – a muted trumpet for four bars – and I'll ask "Did we catch that?"

Most cases I allow other people to mix following my notes. I live in Europe, too, so I'll send a mix back until it's done the way I want to hear it. There are certain key things I want to mix.

HC: You recorded some material at Advision in London. How would you compare recording jazz in Europe vs. the U.S.?

NG: I haven't done too much studio work in Europe. I did some stuff years ago at Barclay's in Paris, the Oscar and Dizzy stuff you mentioned, and I may have done something in Copenhagen with Ella. The session in London was special because of the logistics of the musicians being there. But I didn't find big differences.

HC: I know there was also an OSCAR PETERSON IN RUSSIA set. Was it difficult to record there with government or recording equipment limitations?

NG: I simply had the sound man of the house P.A. record the concert. I don't know what equipment he used.

HC: You seem to have a good rapport with your artists.

NG: I'm very, very comfortable with my musicians and they with me and most of them are musicians I've worked with for many, many years. We have almost a kind of telepathic association where I can say to Dizzy, "Why don't you do a little something there," and he knows what I mean. It's rare that the musicians and I don't communicate immediately.

HC: You mentioned before that you may defer to an artist. How much of a free-hand or authority might be forthcoming from you in the studio?

NG: I've got a session coming up with Milt Jackson and Monty Alexander's trio. I haven't the vaguest idea of what Milt's going to do. He's going to come down and do what he's going to do. At some point I know he'll ask me what I think about it or ask me about tunes he's not sure about chosing. Or I may suggest songs. On the other hand he may have his show all picked out and that's fine. But I would never say to Bags (Jackson), "That's terrible. I won't use it." I would never do that – ever.

If Bags came to me and asked, "Do you like BODY AND SOUL?" I might say, "I hate the tune, but if you want to do it – do it." If he did it, I might suggest it sound better if the tempo were changed, or instead of him opening up, let the piano take a chorus, then come in or walk the chorus – just him and the bass do something. These are just natural suggestions. After all, I've supervised over 20,000

"Rock . . . I'm not saying it's good or bad – but t
is there – I don't think jazz is an art of conformity – and I don'

sides.

HC: Could you comment on the following LP's: The DIZZY GILLESPIE BIG 4 LP.

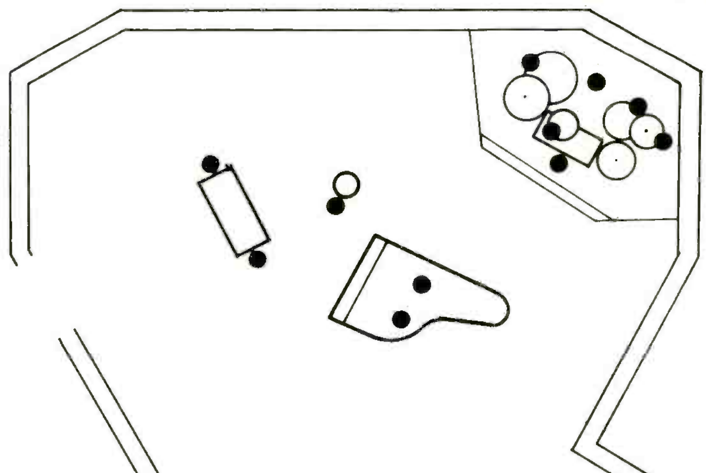
NG: We did that at Cherokee (L.A.) with Ed Greene (engineer). I was very unhappy with that room because it was small. It was a quick thing I had to do with Dizzy because he was there for the day and I ran into the next studio and did Joe Turner with trumpet players right after Dizzy's session.

HC: HAWTHORNE NIGHTS – Zoot Sims.

NG: That was at RCA – L.A. Those were arranged things – Bill Holman (arranger, conducter, leader) wrote the arrangements. I used Bill to do an album with Basie in New York and we discussed doing a thing with Zoot because Zoot, up to that time, had been just doing jamming things. Bill brought his own people in. I don't think I agreed with all of his musical choices. Some of the people he chose – I would have chosen others, but again I deferred to what Bill wanted. What Bill and Zoot were content with, that was fine with me.

HC: ZOOT SIMS AND THE GERSHWIN BROTHERS.

NG: That was easy, again RCA, New York. We did that in one afternoon with people like Oscar Peterson and Ray



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CLIENT Pablo Records
ARTIST/PROGRAM Milt Jackson w/Monty Alexander Trio
ENG. ________ 2nd ________ SET-UP ________

F.B. 2
F.B. 3
F.B. 4

onformity
how it can renew itself."

Brown without arrangements. I called Zoot and gave him a list of tunes he ought to look up.

HC: OSCAR PETERSON – DIZZY GILLESPIE

NG: That was spread over two days because Dizzy wasn't too pleased with some of the things that went down the first day. Again a very casual thing.

HC: ROY ELDRIDGE AND JOE PASS AT MONTREUX – 1975.

NG: There they all got together just immediately before the session. All I did was tell the musicians, "You've got 45 minutes and I think you ought to do at least 4 or 5 numbers." Then they all sat down and decided what they wanted to play. I made some suggestions then they chose the tunes they wanted.

HC: BASIE AND ZOOT (Grammy Winner – 1976).

NG: That began as a Basie trio. Basie came into RCA – New York and we did a couple of sides and he asked me what I thought about having a horn. So I said, "If you want a horn, let's get someone. Let's try Zoot." Zoot came in and sat in on the rest of the session.

HC: What sort of future do you see for jazz?

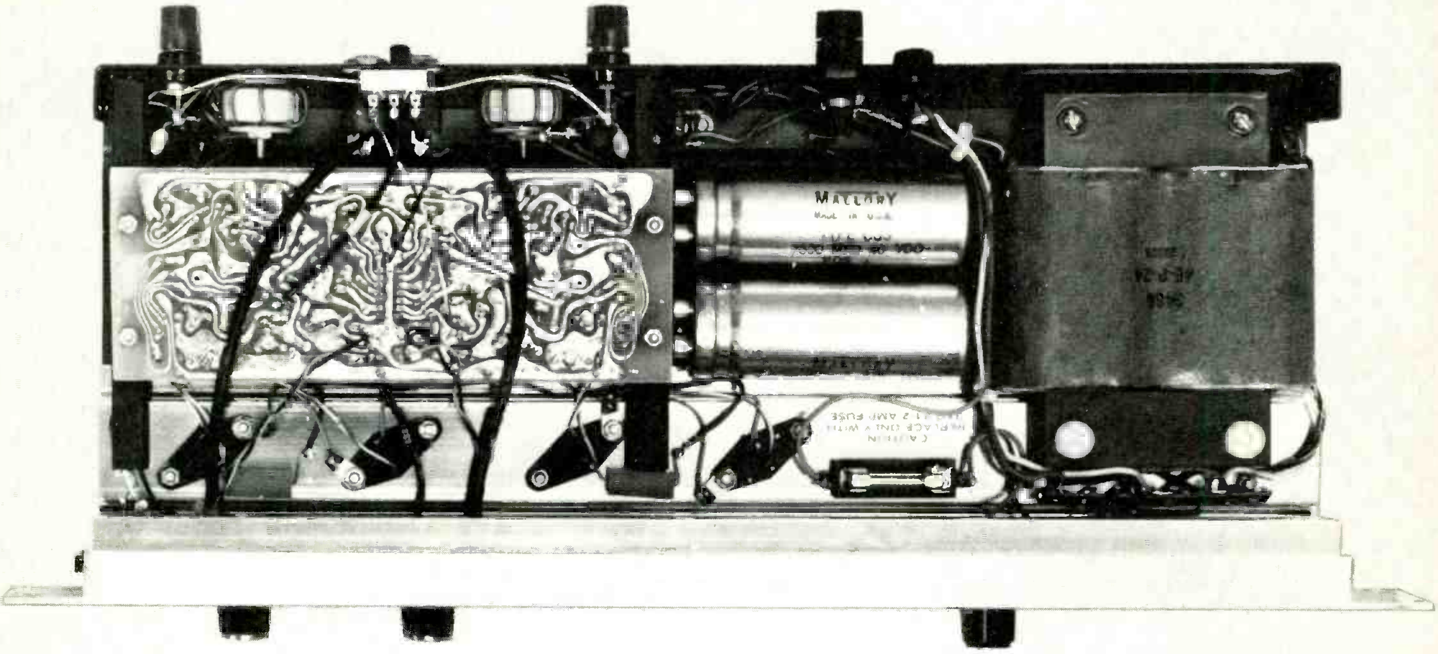
NG: I don't regard so-called cross-over jazz as jazz. I don't think George Benson, who's a very good guitar player, is playing jazz when he's doing BREEZIN' as a single, even though he's jazz-rooted. Nat Cole doing NATURE BOY wasn't jazz even though he was a great jazz pianist.

HC: Do you see your type of jazz renewing itself from younger stock?

NG: I would doubt that jazz is going to survive any more than any of the great individuals that are in any art form. Once they die, I doubt that that type of art could be repeated. I don't think we'll ever have another Picasso – not because he was a great man – the thing that bred him, the environment, just as the environment that bred Lester Young or Roy Eldridge, is no longer around.

I can't see a young musician today, no matter how facile he is technically, being able to make the mark these people did.

For example, you listen to Polish rock and it has all the externals of American rock. I'm not saying it's as good or bad – but the conformity is there. I don't think jazz is an art of conformity and I don't see how it can renew itself. □□□



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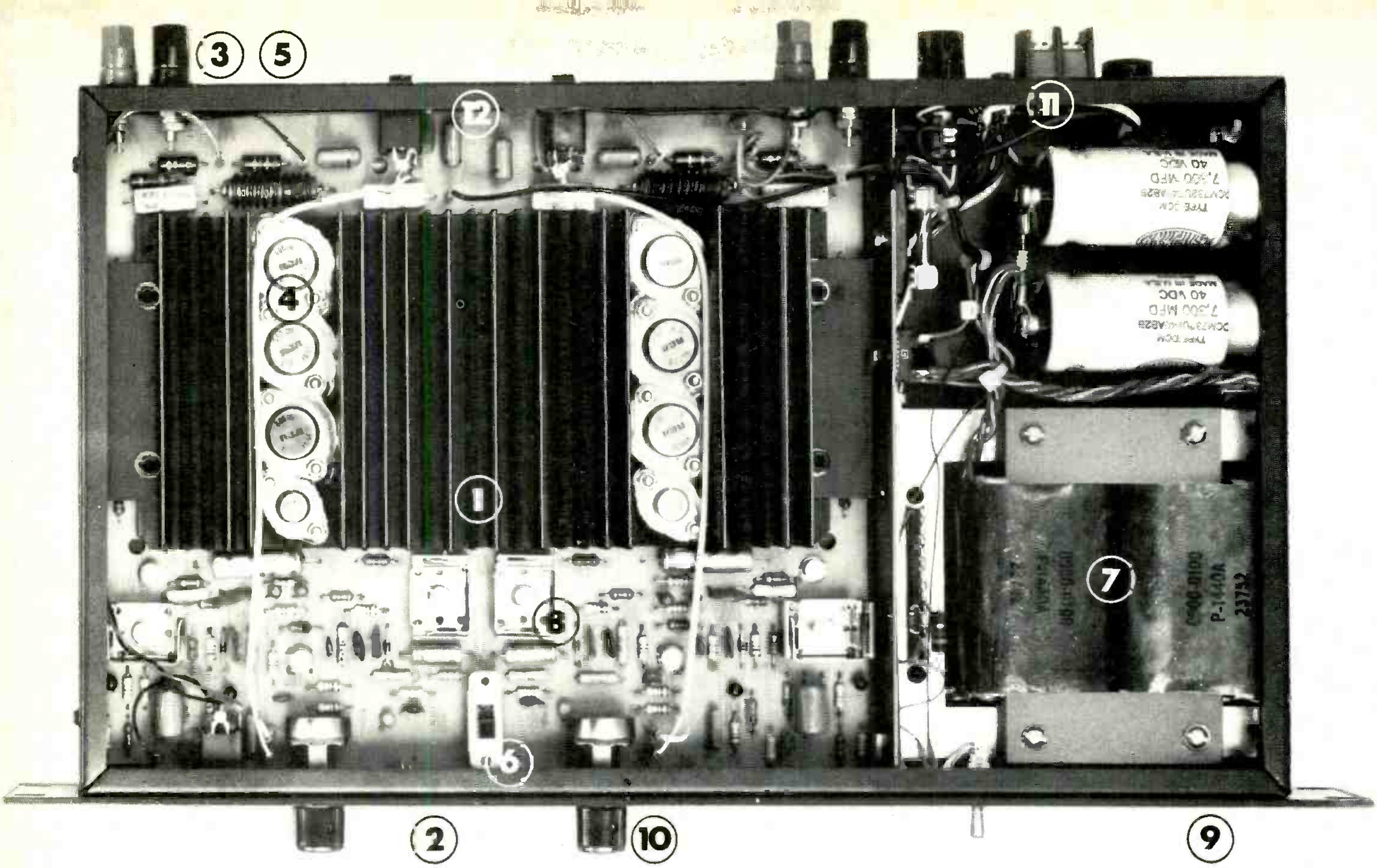

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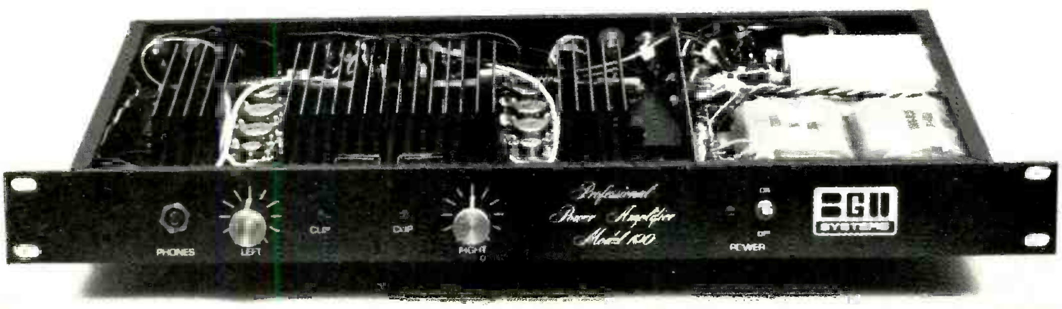

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"No one believes the results of calculations except the man who makes them; everyone believes the results of measurements except the man who makes them."
— Anonymous

Evaluating Loudspeaker Specifications,

Tests, Measurements, Presentations

by CHRIS FOREMAN

A loudspeaker is a transducer, which changes electrical energy into acoustic energy. While basically a simple device, a loudspeaker's actual operation is complex. The world was far simpler when the only criteria for a loudspeaker were "how many inches is it?" and "how many watts does it put out?". In this day of consumer awareness, however, users are beginning to demand more detailed specifications, whether they are buying a component or a complete loudspeaker system. Increasing consumer awareness should benefit the consumer and manufacturer, but it simultaneously places a burden on both: on the manufacturer to measure and publish accurate and useful specifications and on the consumer to know enough to be able to interpret and use them.

The following discussion is an attempt to put some of the more common loudspeaker specifications into perspective. While the emphasis is mainly on specifications that apply to individual components, most of these same specifications apply to speaker systems as well. Since my own background is primarily as a consumer, I write from that viewpoint. For the manufacturer's viewpoints, I am indebted to Walter Dick, who gave me a very informative tour of the facilities at JBL, Manny Mohageri who gave me some insights about test methods at Emilar, Ed Wheeler of Cetec/Gauss who discussed the problem of loudspeaker specifications from an engineering viewpoint, Jim Long, who talked with me at length about the work being done at Electro-Voice, and to Bob Davis and Mark Engebretson who helped me to understand the philosophies behind loudspeaker measurements at Altec.

GENERAL FACTORS AFFECTING MEASUREMENT

1. Where did the measurement take place?

It is often assumed that the only valid loudspeaker measurements are made in an anechoic chamber. However, equally valid measurements can be taken in a number of different situations. The results of measurements made outdoors in a pit or in a reverberant room may differ from an anechoic measurement, but as long as the data taken in one environment can be confirmed by measurements made in another similar environment, we can assume that the original data was accurate.

When measurements are done in an anechoic chamber, a measuring microphone placed in front of the loudspeaker will not be affected by energy radiated from the sides or the rear of the loudspeaker. If the loudspeaker is placed in a pit flush with the ground and the microphone is placed on a crane above the pit, some energy from the rear of the loudspeaker may be reflected forward to join the sound from the front of the loudspeaker, reaching the microphone at nearly the same time. How much of the rear energy reaches the front depends on the "frequency response" and "reflectivity" coefficient of the material surrounding the pit. For a highly absorbent pit, the reflections are typically 40 dB below the direct sound level so that measurements are similar to those taken in an anechoic chamber. Occasionally, certain measurements are made in a reverberant room. These will be different from either anechoic or outdoor-pit measurements.

None of these environments has any inherent, overwhelming advantage over another for measurement accuracy or usefulness. In addition, without a lot of data about the environment itself, only gross comparisons can be made between loudspeaker specifications taken in different environments. The important thing to recognize is that the measuring environment has a lot to do with the numbers that get published.

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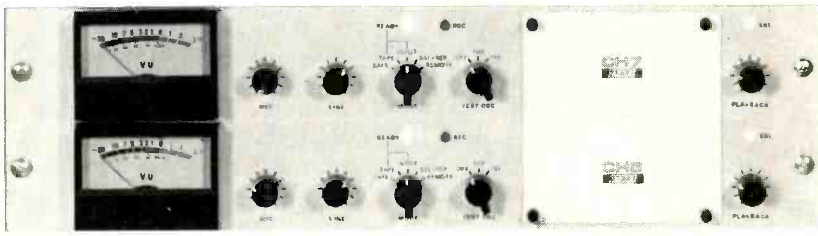

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speaker evaluation

2. What is the physical layout for the tests?

The physical positioning of the loudspeaker and test microphone can significantly affect the results of measurements. As an example, most "distribution" measurements are done with the loudspeaker placed on a turntable, or the microphone on a tiltable crane. This seems straightforward enough provided that the center of rotation is defined. Figure 1 illustrates several possible positions for a loudspeaker on a turntable or a microphone on a crane. If the horn driver in this example is the axis of rotation, the results of a polar plot may be very different than the results for the same driver/horn combination when the mouth of the horn is taken as the axis of rotation. A center axis may be the best compromise, yet it is not perfect. At low frequencies, the mouth of the horn appears to be the acoustic source. At high frequencies, the diaphragm of the driver appears to be the acoustic source. Thus, a center axis can be somewhat misleading, too.

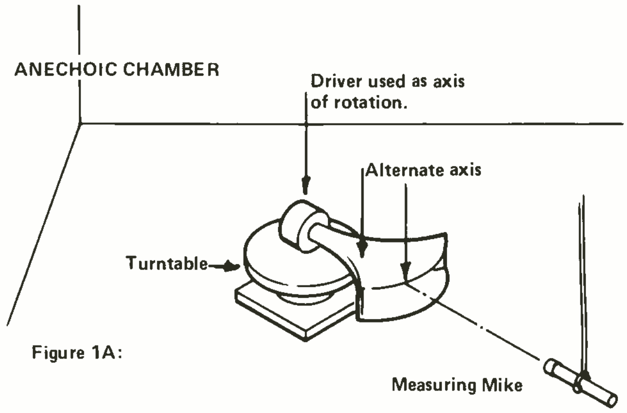


Figure 1A:

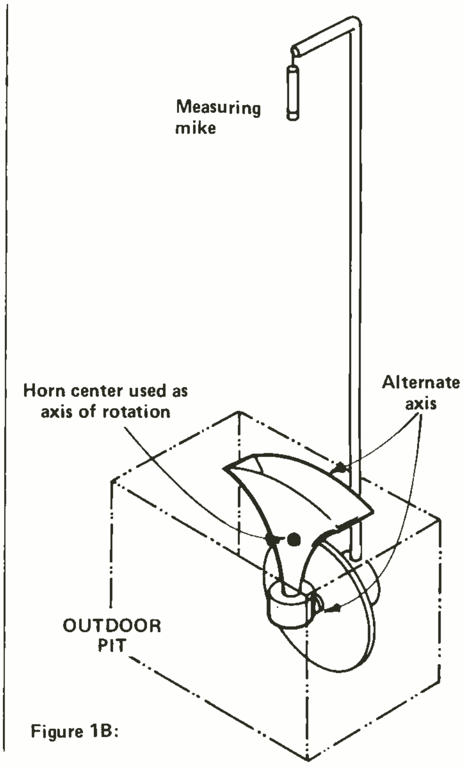


Figure 1B:

Figures 1A & 1B: Possibilities for axis of rotation in polar measurements.

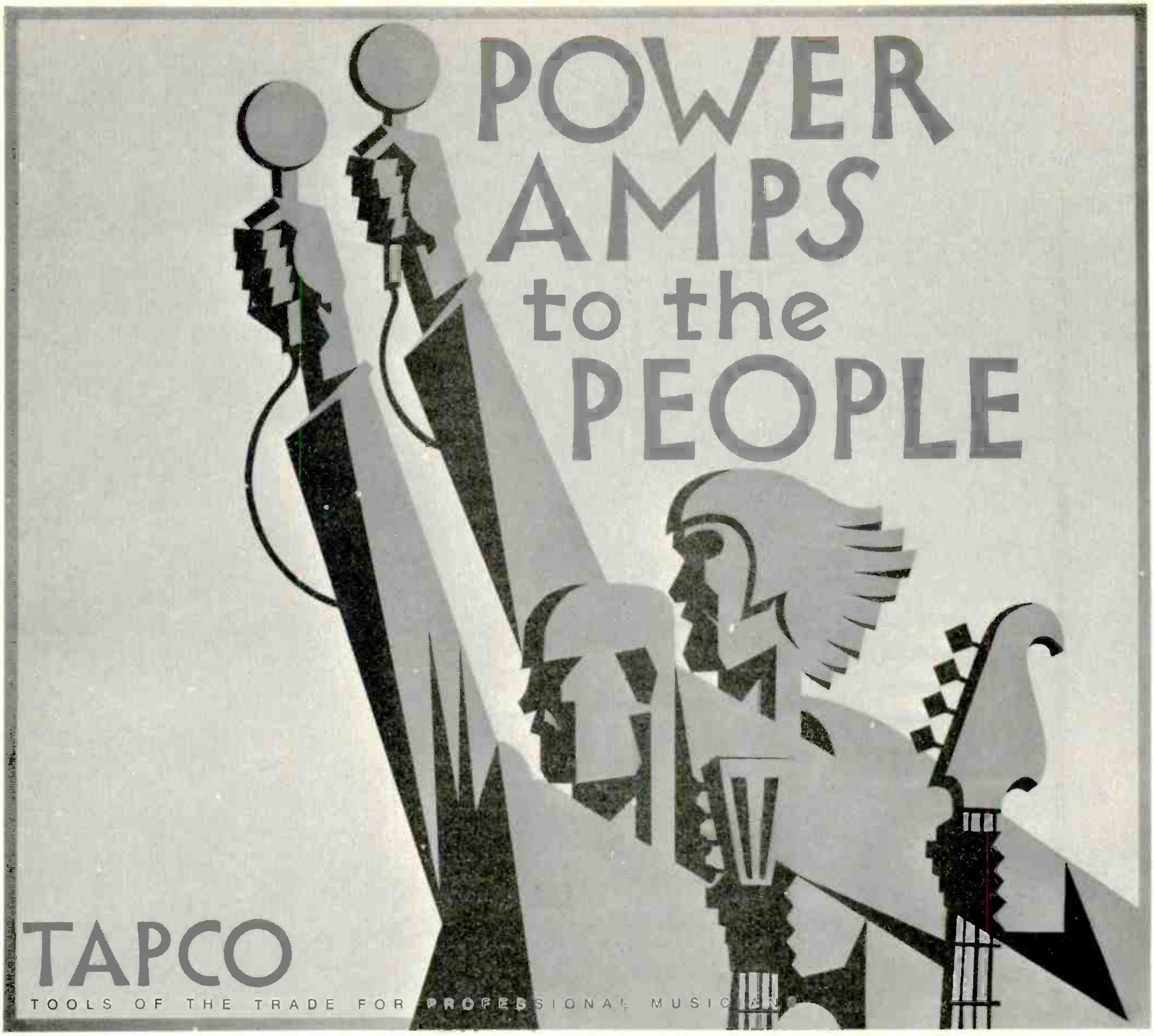
One way to minimize this problem is to place the measuring microphone a considerable distance away from the loudspeaker. This way, any shifting of the apparent acoustic source is a small fraction of the actual measurement distance. This works great if you have a giant anechoic chamber or a perfectly quiet outdoor measuring facility.

Another layout factor involves the distance, and the resulting time delay, between the test loudspeaker and the measuring microphone. If the test equipment is not carefully compensated, this can cause significant measurement errors during some tests.

3. What is the acoustic loading on the component under test?

When a high frequency driver is under test, the horn (or plane wave tube) it is connected to affects the results. Similarly, the enclosure for a cone loudspeaker affects the way it performs. Measurements such as polar plots and "Q" can only be made on a combination horn/driver or cone-loudspeaker/enclosure. But measurements of frequency response or total efficiency depend on the horn/enclosure, too. Except for musical instrument speakers, which may be installed in an open backed guitar amplifier, most component loudspeakers will not perform properly unless they are carefully loaded acoustically. The horn or enclosure acts to load the component loudspeaker in a manner similar to "terminating"an impedance-sensitive delectronic device.

4. What is the source material used for the test?



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speaker evaluation

Pink noise, white noise, warble tone, sine wave, spot frequency, program material -- which is the best? Unfortunately, there's no clear-cut answer to this question either. Pink noise seems to be gaining popularity for a number of tests. Pink noise has an equal amount of energy per octave or fraction of an octave. Therefore, the amount of energy between 100 Hz and 200 Hz, a 100 cycle wide segment of the audio spectrum, is the same as the energy between 10 kHz and 20 kHz, a 10,000 cycle wide segment. If the peak to average ratio of a pink noise signal is held to approximately 10 dB, the noise corresponds roughly to the average energy content of music. This is one argument for the use of pink noise. Pink noise signals are often derived by filtering white noise signals. White noise has an equal amount of energy per Hz, which means that the energy between 100 Hz and 200 Hz is equal to the energy between 10,000 Hz and 10,200 Hz. Both pink and white noise are useful in non-anechoic measurements since they randomize the effects of room reflections, and can provide a constant bandwidth test signal. Lately, warble tone tests have become less popular and have been replaced by noise tests since noise provides similar advantages and is more easily specified.

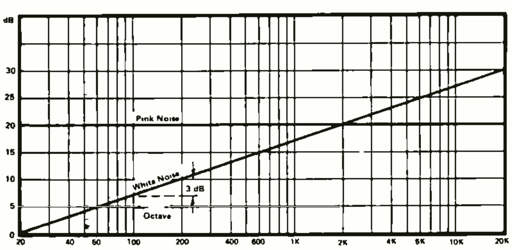

Figure 2: White Noise and Pink Noise on a Logarithmic Frequency Scale

A sine wave source is useful in some tests. Frequency response graphs are often made using a sine wave that is swept throughout the frequency spectrum under consideration. Spot frequency tests are made using single frequency sine waves at selected points in the sprectrum. While swept sine wave tests are informative, spot frequency tests may hide significant anomalies in a component's response.

5. What are the production tolerances? How representative is the test unit of actual devices being sold to the user?

This is one factor that is often ignored in a manufacturer's specifications. In the back of our minds, we all know that there are differences in new components with the same model numbers, but very little information is available to tell us the actual magnitude or statistical distribution of such differences. In addition, we seldom know how the component used for the tests was selected. Is it a hand-assembled prototype or a randomly selected production unit?

Transistor specification sheets commonly show "maximum", "minimum" and "typical" values for a given specification. This could be one valid way to rate loudspeakers. Another possibility would be to "grade" loudspeakers similar to the way some transistors are graded. Because of the way transistors are manufactured, the differences between maximum and minimum values may be 100% or more. Thus, some transistors are sorted into categories that indicate whether they are closer to the maximum or minimum value for a particular specification. A 2N2926, for example, can be purchased with several different color codes which identify current gain gradings. The higher the current gain, the higher the price. Some microphones are already tested and graded in a similar manner. A given microphone cartridge is placed in a "premium" line microphone or in an "economy" line microphone according to how well it fits a pre-determined set of criteria. However, grading loudspeakers according to smoothness of frequency response, or according to overall efficiency would be an expensive process for the manufacturer and therefore for the consumer. Thus, any decision to do this would have to be based on practical as well as theoretical matters. Still, it would be useful to at least have some idea what typical manufacturing tolerances are.

6. Paper factors.

The way a specification is presented on the final spec sheet affects the way it is interpreted. As an example, consider the two frequency response plots in Figure 3. The vertical scale of Figure 3A makes it look like a much smoother graph than that of Figure 3B, even though the two graphs represent very similar devices.

Another example of a "paper factor" is the ± so many dB rating. A frequency response that is "flat" within ±3 dB from 100 Hz to 15 kHz may be "flat" from 50 Hz to 20 kHz if the ± rating is increased to ±6 dB. In addition, the reference frequency chosen as the "0 dB" level should be specified.

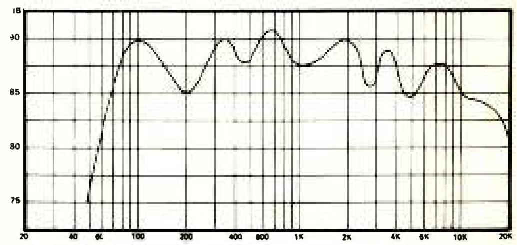

Figure 3a: Expanded Scale Frequency Response

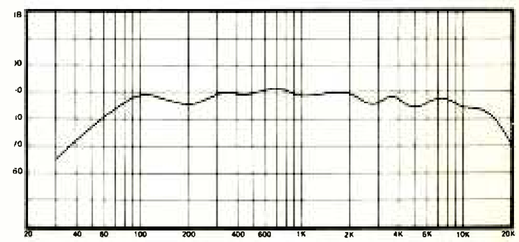

Figure 3b: Compressed Scale Frequency Response of Same Speaker as 3a

One often overlooked paper factor is a manufacturer's decision of just which specifications to publish. It must be tempting to publish those specifications that make a product look good, and to "forget" to publish any that make the product look bad. In a business where specmanship is the name of the game, it is probably safe to say that this practice exists. There are other reasons, however, for not publishing certain specifications. One is that, at least in the past, we as consumers would have been thoroughly confused by some specifications such as comparisons between "distribution angle" and "Q" for a given speaker. This data is only useful to someone who knows how to use it, and thus, has often been available only on special request.

One final paper factor is the publication of the so-called "one-number" specs. A power capacity rating of "100 watts", a frequency response rating of "30 Hz to 18 kHz" or a sensitivity rating of "93 db" are examples of "one-number" specs. Without supporting information each of these specifications is suspect at best. A more informative power capacity spec, for example might read something like this: "100 watts of pink noise, with a

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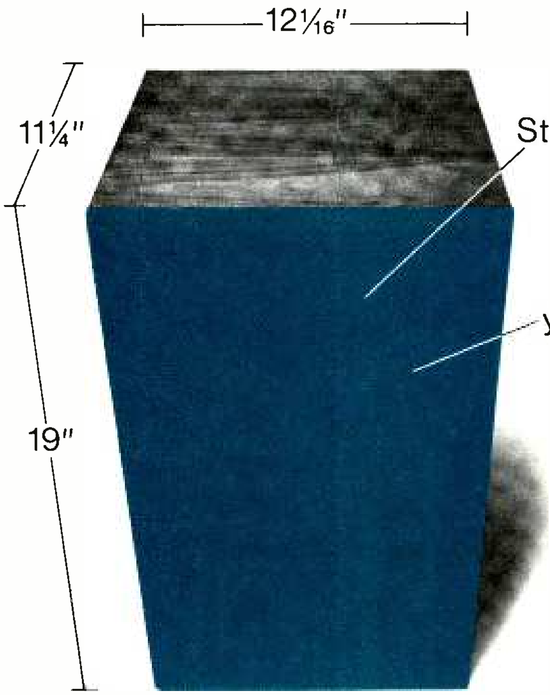

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SPECIFIC SPECIFICATIONS

1. Frequency response.

The frequency response of a component loudspeaker is a plot of the acoustic output versus frequency for a given electrical input. Generally, the quantity measured is *acoustic amplitude,* measured in SPL. Occasionally, acoustic *power* output is substituted for acoustic amplitude. Many driver tests made on plane wave tubes are actually power frequency response tests, not amplitude frequency response tests. The difference is important. A given high frequency driver may have a power frequency response as measured on a plane wave tube that looks something like Figure 4A. When placed on a horn that is somewhat "beamy",* the apparent amplitude frequency response improves to that shown in Figure 4B. This does not, however, imply that plane wave tube measurements of a driver's frequency response are necessarily more "valid" or even more informative than measurements made on a high frequency horn. First of all, there are no published standards for "plane wave tubes", and since different manufacturer's drivers often have different sized throat connections, this would seem to be a difficult standard to derive. Secondly, the driver will be used on some kind of horn in the field, so measurements made on a horn offer helpful information to the user.

In an anechoic chamber, frequency response tests may be made using a sine wave swept through the audio spectrum. This should give a plot that shows up any and all faults in the component's frequency response. In a non-anechoic environment, pink noise is often substituted as the source material, in order to randomize any reflections. Plots are then given in terms of third-octave, half-octave or full-octave bands. Since pink noise plots are a common way to verify frequency response of a completed system in the field, a manufacturer's pink noise plots provide helpful information to the user. However, it's possible that a pink noise plot showing only full-octave bands may not display sharp peaks or dips in a loudspeaker's response.

One method that should always raise questions in the user's mind is that of "spot frequency" tests. The equipment needed to do spot frequency tests may be less expensive than the equipment needed to do either pink noise or swept sine wave tests, but unless a large number of spot frequencies are plotted, the results can be very misleading. Figure 5 shows what could happen if a loudspeaker with severe peaks and dips in its response were measured using a spot frequency plot with "conveniently" placed frequencies.

2. Polar pattern and dispersion.

Usually, frequency response plots show a loudspeaker's "on-axis" response. Occasionally, several plots are given on the same graph to show the frequency response at various horizontal and/or vertical angles away from the on-axis response. Another way to show how a loud-

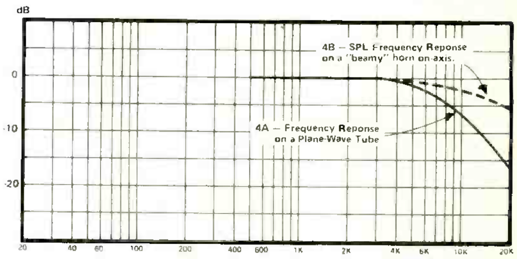

Figure 4a & 4b: Frequency Response of a Typical High Frequency Compression Driver

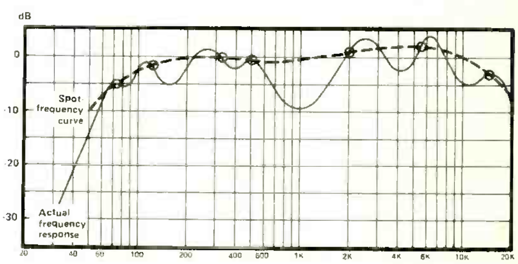

Figure 5: Misleading "Spot Frequency" Frequency Response Curve

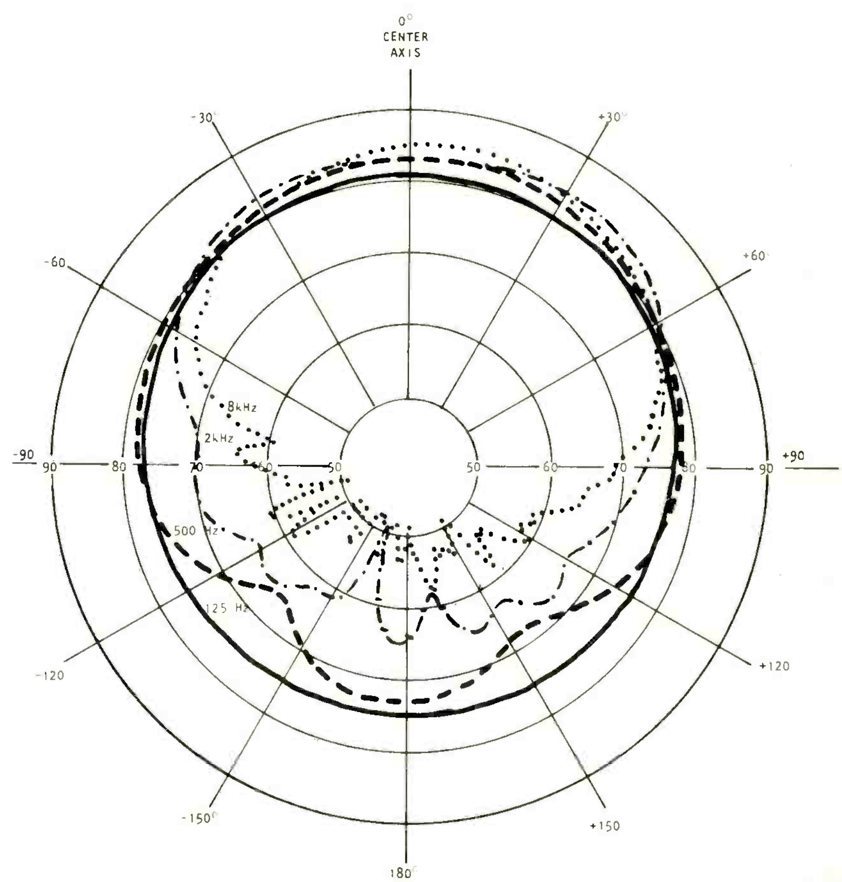

Figure 6: Polar Plots

* *Almost all horns display some narrowing at high frequencies.*

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ACOUSTIC RESEARCH "FINGERPRINT TEST" REVEALS HIDDEN TRANSIENT REPSONSE DETAILS

Scientific tests developed in recent years allow loudspeakers to be measured in two ways. One such method, which provides a unique "fingerprint" of every speaker tested, is the *transient decay plot.* This method requires a digital computer and plotter, and shows how truthfully a speaker system reproduces a transient sound at different frequencies. AR is one of the few high fidelity companies in the world with full in-house facilities for making such plots.

Figures 1 and 2 show how two different loudspeaker systems reproduce a single impulse, such as a pistol shot. This type of single impulse contains all frequencies, from 0 to 20,000 Hz at constant amplitude. A perfect reproduction of this would be a single flat response curve. The beginning of each plot is at the top; at the end of the plot, (at the bottom of the page) it is 1/200th of a second later. Each fine line is a separate frequency response curve, delayed fifty millionths of a second from the one before it.

The speaker in Figure 1 has a very uneven response, with a persistent resonance at about 13,000 Hz. The energy comes out slowly, at all frequencies, instead of rapidly, as would be the case for an ideal loudspeaker.

Figure 2 shows a system which has a very smooth response over the entire frequency range, followed by a rapid decay, and is relatively free of peaks.

By showing the detailed effects of small changes in design, such computer techniques help AR engineers develop better-sounding speakers. This is one of many applications developed in AR's research computer laboratory.

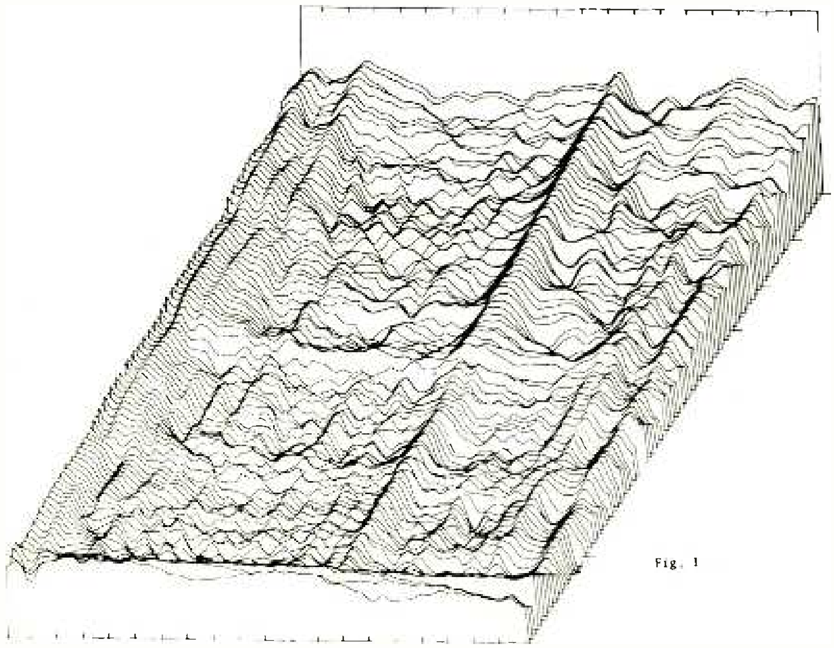

Fig. 1

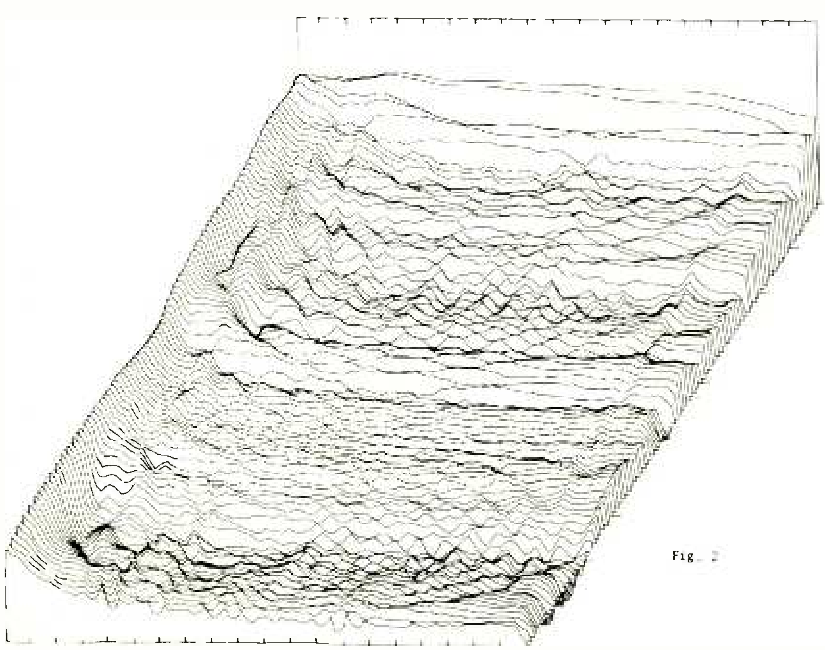

Fig. 2

speaker distributes its acoustic output is by "polar plots". To make a polar plot, the loudspeaker is placed on a turntable in an anechoic chamber with a fixed microphone some distance away. The turntable's rotation is synchronized with a chart recorder that plots data on a circular graph paper, thus giving a plot of acoustic output versus angular position around the loudspeaker. Alternately, for an outdoor pit measurement, the measuring microphone may be placed on a tiltable crane and rotated around the loudspeaker. Normally, a single-frequency sine wave is used for the test, and several tests are done at various frequencies giving a "family of curves" such as that shown in Figure 6. This method has some of the same dangers as the spot-frequency/frequency-response tests. A swept-sine wave polar plot, however, would produce a three-dimensional graph, and would be difficult to measure or display. Pink noise may be used to "average" out the response over a given frequency band.

Polar plots should be displayed in both horizontal and vertical directions. When both horizontal and vertical plots are given for a number of different frequencies, we have a reasonable picture of the angular coverage of the loudspeaker.

The assumed location of the acoustic source can affect a polar plot. Again, the best way to avoid this problem is to place the measuring microphone as far away as possible from the loudspeaker under test. It would be valuable, in any case, for the manufacturer to publish the test methods as well as the test results of polar plots.

3. Directivity: Q or R.

The Q of a loudspeaker relates its on-axis sound pressure level to what that level would be if the loudspeaker were a perfect spherical source radiating energy equally in all directions. A high Q value implies that the loudspeaker radiates most of its energy in front, and little energy from its sides, top, bottom or back.

The Q of a loudspeaker can be determined graphically from its polar plots. Sometimes a "horizontal Q" and a "vertical Q" are defined, calculated from horizontal and vertical polar plots. A "mean Q" can then be derived from the horizontal and vertical Q values. This mean Q approximates true spherically measured Q. There are certainly other methods for measuring a loudspeaker's Q, but, again, there is no set standard, and comparisons between manufacturers devices on the basis of published Q values is risky at best.

For the user in the field, however, ac-

curate Q data is very useful. The sound radiating from the sides, top and bottom and rear of a loudspeaker is seldom useful, and can contribute to the reverberant field in a room without adding any intelligibility. Thus, for speech reinforcement, in a highly reverberant room, a high Q loudspeaker is superior to a low Q loudspeaker.

It should also be noted that two loudspeakers with the same ±6 dB coverage angles can have very different Q values. Thus, especially in the absence of families of polar plots, Q is a useful description of a loudspeaker's performance.

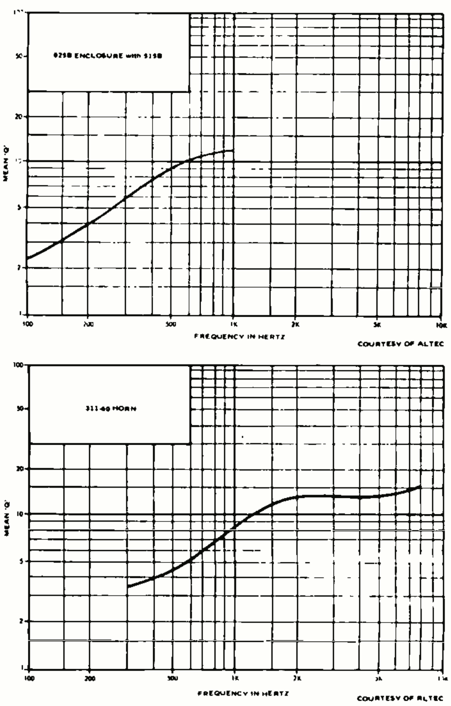

Figure 7: 'Q' Plots for two Loudspeakers

4. Sensitivity.

Sensitivity is measured by placing the measuring microphone on axis at a specified distance from the loudspeaker. The sensitivity is the SPL measured at the microphone for a given electrical input to the loudspeaker.

Generally, the specified distance is 4 feet although 1 meter is becoming more popular. Sometimes, the measuring microphone is placed farther away from the loudspeaker, and the SPL at 4 feet or 1 meter is calculated via the inverse square law. This is a good tactic for a speaker system (such as a studio monitor) since it gets the microphone well away from the loudspeaker's "near field", where measurements might be inaccurate. The EIA (Electronic Industries Association) specifies a standard sensitivity test that uses a distance of 30 feet. In a small anechoic chamber, this test may be done at a shorter distance with the results calculated again by the inverse square law.

Like most other specifications, source material affects the sensitivity measurement. A pure sine-wave at a single frequency can give a valid sensitivity rating, although its value is limited to that particular frequency. If the single frequency chosen happens to be coincident with a severe peak in the loudspeaker's frequency response, the sensitivity measurement will appear much better than could be realized under actual user conditions. A swept sine wave sensitivity plot gives us the same information as a frequency response plot except that the curve height is calibrated. Pink noise is a useful way to average the sensitivity value over a specified bandwidth. When a pink noise sensitivity rating is given, the pink noise bandwidth should also be given. White noise sensitivity ratings are occasionally used, and may give higher or lower values than a pink noise rating on the same loudspeaker.

One of the most important considerations in sensitivity ratings is the power level fed to the loudspeaker. Most often, the power level is specified as "1 watt" (EIA specifies 1 mW). However, the method of calculating this watt may or may not be specified. Power can be calculated by squaring the voltage input and dividing by the nominal (rated) impedance of the loudspeaker. This is probably the most popular method, but can be misleading if the actual impedance of the loudspeaker drops significantly below the rated nominal impedance. A lower impedance value will raise the actual power delivered to the loudspeaker, increasing the apparent sensitivity value. Thus, for sensitivity, a plot of impedance versus frequency would be useful, or alternately, an explanation of the method of rating impedance, or at least a nominal and minimum impedance value.

Another way to calculate the power level is to multiply the actual voltage delivered to the loudspeaker times the actual current. This automatically compensates for impedance and keeps the power delivered at an accurate value. It should be a good method for sensitivity measurements made with band limited pink noise since the impedance value would be effectively averaged over the frequency band under consideration.

The proposed AES loudspeaker measurement standards may suggest that sensitivity be given with power input specified in terms of so many volts. This would eliminate the problem of how to specify power since, at a given voltage, the power level delivered to the loudspeaker is determined by the speaker's impedance. It would be a reasonable method because power amplifiers don't really deliver power, they deliver voltage; their power output is determined by the loudspeaker's impedance. This method would, however, make a 4-ohm loudspeaker seem more sensitive than a similar 16-ohm loudspeaker.

Sensitivity values are also affected by the apparent acoustic source. For a horn/driver combination, the sensitivity will seem to be higher if the apparent source is the mouth of the horn that if the apparent source is the driver diaphragm. As with polar plots, the measuring microphone should be located as far as possible from the loudspeaker to minimize errors from shifting of the apparent acoustic source. In addition, a "4 foot" (or 1 meter) rating should specify 4 feet from *where* (mouth, diaphragm, cone center, etc.).

For a loudspeaker/enclosure or driver/horn combination, the sensitivity value is significant and useful. Along with the frequency response and horizontal and vertical polar plots, sensitivity allows useful calculations of the actual SPL at a given distance and angle from the loudspeaker in an actual user environment (provided, of course, that the environmental conditions are also taken into account). A single component sensitivity, as for a high frequency driver or a low-frequency cone loudspeaker is suspect, however, unless the enclosure/horn used in the measurement is also specified. For a driver, a sensitivity measurement on a plane wave tube may provide very different results than for the same driver on a wide-angle horn. Similarly, for a low-frequency cone speaker, a senstivity measurement made with the speaker in a carefully designed and tuned enclosure will probably lead to better results than for the same speaker mounted in a 5 cubic foot infinite baffle. What is important here is not which measuring method is best, but that the conditions under which the test was made be published along with the results.

5. Power capacity.

If there's a real bug-a-boo in this whole

Nashville's Hottest Group

Claude Hill, the rotund lead vocalist for Audio Consultants was recently asked why he felt his group has had such phenomenal success. To quote Claude: "In addition to giving our public all the standards we have come to know so well, my group has continually worked out new material. You know, fresh approaches to the same old tunes. I'm no prima donna either. I couldn't do it alone. I count on everybody in Audio Consultants to hold their own. It's real team work."

"Regardless of the gig, after we perform there's a real sense of joy—a job well done."

Well said, Claude. Like anything worth waiting for, we'll just have to see if this group can hold on to their position at the top. They know sound like no other group in the business and if what they've already achieved is any indication of the future, then keep an eye (and an ear) on Claude & his Audio Consultants.

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Don Woerner is the man in Dallas.

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specifications mess, it would have to be power capacity measurements. The power capacity of a loudspeaker is very different from the power capability of a power amplifier. A loudspeaker's power rating describes its ability to *absorb* electrical power which is converted into acoustic power and waste heat. A power amplifier's power rating describes its ability to *produce* electrical power and to deliver it to a specified load. Yet these two different ratings have the same unit, the watt. This causes a confusion whose symptoms are displayed by questions like "How many watts will this speaker put out?" (with the answer expected in electrical, not acoustical, watts).

Basically, a loudspeaker's power capacity rating is supposed to help us determine the maximum acoustic output from the loudspeaker over a given frequency range and time period. It should also help us determine the largest power amplifier we may safely connect to the loudspeaker. Yet a single rating, even if measuring conditions are specified, is not fully descriptive of the actual loudspeaker power capacity.

Loudspeaker failures can be divided into mechanical and thermal types. Mechanical failures usually happen when too much power is applied at a low frequency, such as a high power transient, causing over-excursion of the cone or diaphragm. Mechanical failures can also happen from simple material fatigue after an extended operational lifetime. Thermal failures usually result from too much average power applied over a broad range of frequencies. Figure 8 is a graph showing the mechanical and thermal power limits for a typical high frequency driver.

Power capacity ratings are affected by the source material used in the test. Pink noise, white noise, sine wave and "program material" are typical sources. Since what we usually want from a power capacity rating is some idea of how the loudspeaker will stand up under normal usage, the program material source might seem to be the most suited. However, it is very difficult to define "program material", so this power capacity test is largely unrepeatable. Pink noise, when its frequency range and peak to average ratio is specified, constitutes a repeatable source similar to program material. Thus, pink noise power ratings are useful provided all other aspects ot the test are known. White noise power tests should also be repeatable, although they will not give the same results as pink noise tests. In addition, because of its high energy at low frequencies, pink noise is a more severe test for most loudspeakers; in other words, if the same loudspeaker's power capacity were rated with both pink and white noise, the pink noise rating would most likely be lower.

Sine wave tests for power capacity neglect the fact that most program material exhibits a high peak to average power content. Thus, a pure sine wave power capacity is often rated in "RMS watts" even though the concept of RMS does not apply mathematically to *any* kind of power rating. This rating is commonly derived by measuring the loudspeaker's maximum RMS sine wave *voltage* capacity, squaring it, and dividing by the loudspeaker's nominal impedance. This would be similar to a "continuous average sine wave power" capacity for an amplifier, but it does not necessarily apply to loud-

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speakers.

Acoustic loading affects the loudspeaker's power capacity, too. Thus, the horn/enclosure used in the test should be specified. Almost any good horn or any well-designed enclosure that fits the driver/speaker is usable. A plane wave tube may also be used for a driver.

Another factor in power capacity testing is how long the loudspeaker was tested. When a power level is found that the loudspeaker can tolerate over long periods, that level is designated as the rated power. The length of period used for this final test should be specified. One hour and twenty-four hours are common test periods. Again, what is important is that we as users become aware of the various factors involved in power capacity specifications. In addition, manufacturers should publish the conditions as well as the results of these tests to whatever extent is possible.

6. Distortion.

The trouble with this specification is that there are so many different types and varieties of loudspeaker distortions: "phase distortion", "IM", "doppler", "harmonic", "cone breakup", "transient" and so on. Some are truly irritable, and others are tolerable in relatively high amounts. In addition, distortion depends on source material, frequency, acoustic loading (horn/plane-wave-tube/enclosure) and whether the speaker is working alone or in a system. Thus, because of their extreme complexity, distortion ratings for professional loudspeakers are rare, yet in some circumstances, they could be useful.

The harmonic distortion of some loudspeakers increases near their upper frequency limits. Thus, a frequency response curve made with wide range source material, such as pink or white noise, may display a high frequency response that is at least partially made up of distortion components from the lower frequencies in the source material. For this, and for other reasons, a harmonic distortion specification could be enlightening if the measuring conditions were specified in detail.

On the other hand, the terms "harmonic distortion", "intermodulation distortion" and so on, are primarily used for specifying non-linearities in amplifiers (electronic non-linearities). Yet, even though their causes are very different, we tend to interpret loudspeaker harmonic or intermodulation distortion specifications (mechanical/acoustical non-linearities) as if they were electronic in nature. Single sine-wave harmonic distortion ratings for consumer loudspeaker systems are fairly common, but it may be that these are published in order to satisfy those who think that loudspeaker ratings ought to be similar to amplifier ratings. Perhaps what is needed is an examination of the actual non-linearities inherent in a loudspeaker, what types of distortions they produce, and some recommendations of realistic methods of specifying those distortions. Anyone like to volunteer?

7. Other specifications.

Impedance, while not actually a performance specification, is still certainly useful. We need to know the impedance of a loudspeaker to determine how many loudspeakers of the same type can be connected to a power amplifier without overloading the amplifier. In addition, an

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impedance rating is necessary to calculate the power delivered to a loudspeaker when a given power amplifier (with a specified *voltage* output level) is connected to the loudspeaker. Finally, impedance is useful in interpreting several performance specifications, as discussed above. Impedance is often given in graph form along with a frequency response curve. Alternately, a "nominal" (what does "nominal" mean?) impedance is rated, or a nominal and a minimum. It is signficant that if a loudspeaker's minimum impedance falls lower than its rated, nominal impedance, some specifications (notably sensitivity) may appear better than could be realized in practice.

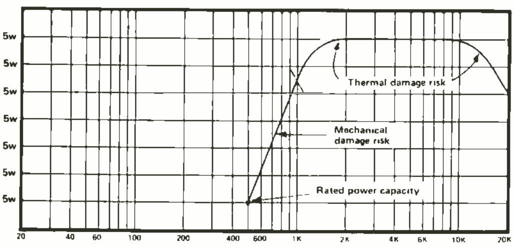


Figure 8: Mechanical and Thermal Power Limits for a Typical High Frequency Driver

Recommended crossover frequency is another useful specification which can give us some idea of the low frequency power limits of the loudspeaker. Normally, the recommended crossover frequency will fall in the region where the loudspeaker's mechanical and thermal power limit curves meet (see Figure 8). In addition, the recommended crossover frequency can be a clue to the loudspeaker's polar response. Outside of their designed frequency range, most loudspeakers' polar response varies widely from the optimum. Thus, a properly specified recommended crossover frequency can help the user stay within a desired loudspeaker distribution pattern.

Free air resonance is sometimes specified, especially for low-frequency cone-type loudspeakers. This can be useful in building certain types of speaker enclosures. A recent alternative is to publish the "Thiele/Small" parameters which allow enclosures to be constructed according to methods given in several AES papers written by A. N. Thiele and Richard H. Small.

Time response specifications seem to be gaining popularity. A time response plot is analogous to a frequency response plot, except that time, instead of frequency is compared to SPL for a given input signal, usually a tone burst or impulse of some kind. Three-dimensional plots are also gaining popularity. These look like mountainous landscapes, and usually are plots of amplitude versus frequency versus time. This is somewhat like a plot of the way a frequency response curve changes with time. The difference is that the input signal to the loudspeaker is not necessarily constant, but may be a tone burst or impulse in place of pink noise or swept sine wave. Time response plots can tell a lot about the transient response of a loudspeaker, and they would be useful in a loudspeaker specification sheet if they could be standardized.

The phase response of a loudspeaker can be indicative of its performance. Since a loudspeaker is a non-minimum-phase device, its phase response cannot be predicted from its frequency response. Thus, a phase versus frequency graph could present new information. Poor phase response, for example large phase shifts with abrupt changes, means that acoustical wave*shapes* will be significantly altered from the electrical waveshapes fed to the loudspeaker. For program material with a high transient content, this alteration of waveshape can apparently be quite audible.

There are a number of specifications which can be lumped together and called non-performance specifications. Physical dimensions are included in this set. So are specifications such as "magnetic weight", "BL factor" and so on. While physical dimensions and overall weight are important specs, magnetic specifications, and various other specifications describing the frame structure or material, or the coil winding technique, etc., are of little use. The space taken up by these specifications might be better used to display Q-plots or to discuss measurement conditons for other specs.

Gated testing.

A gated test, such as those described by several excellent B & K pamphlets, is a method of obtaining "free-field" results in a non-free-field environment. The concept is simple: A tone-burst is the source. The measuring microphone is electronically switched on and off (gated) at just the right time to accept the tone burst output of the loudspeaker but to reject any other sounds such as noises or echoes from walls. One fault with this method is that the room must be large enough that echoes will arrive some time after the tone burst (the direct sound) has arrived. Yet the tone burst must be long enough to complete at least a full cycle of the lowest frequency under consideration. In addition, the gate normally cuts off the leading and trailing edge of the tone burst which means that the loudspeaker's transient reponse cannot be easily measured. Still, under the right conditions, gating can be a useful test method.

Listening Tests.

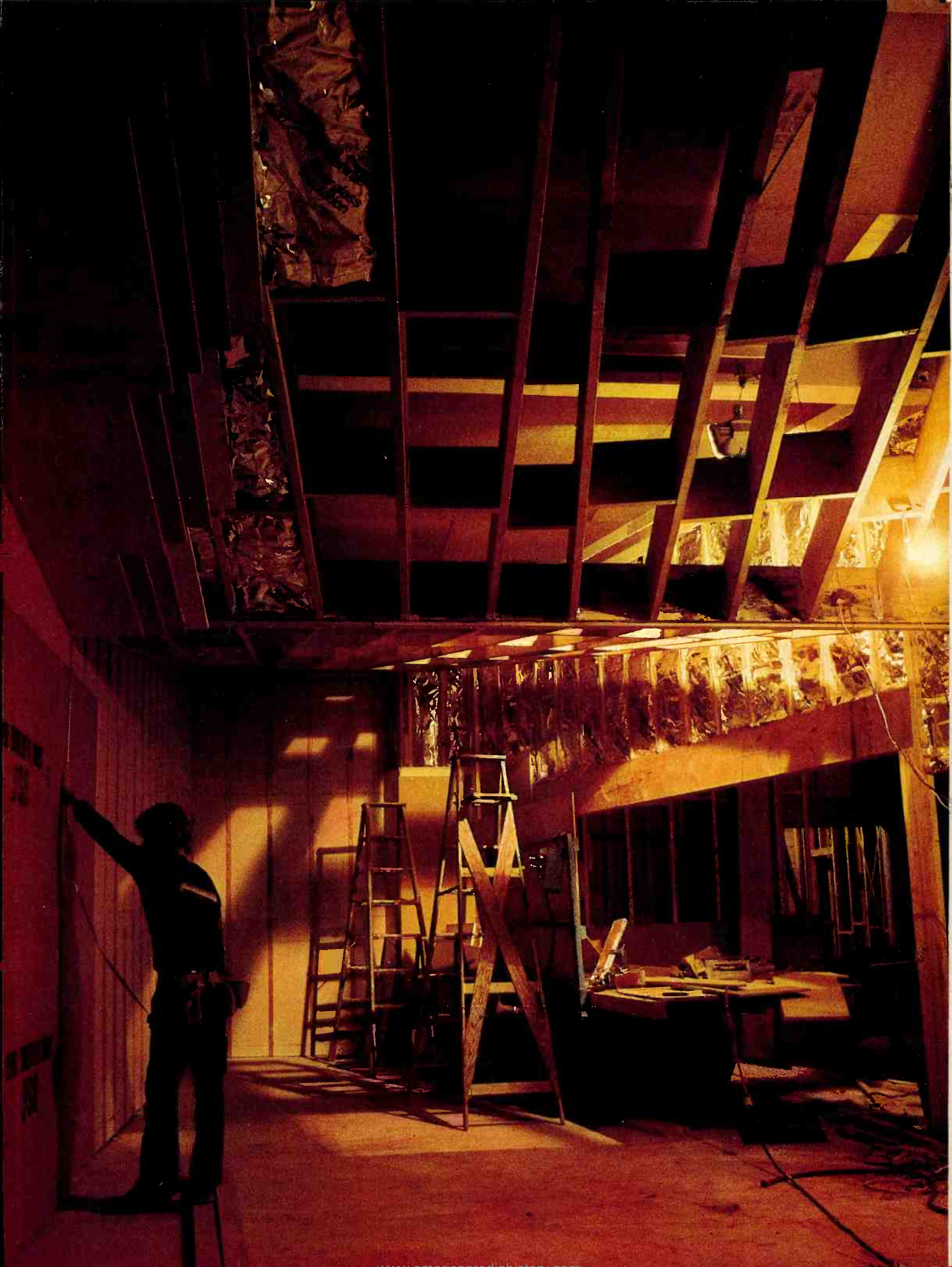
Listening tests allow subjective judgements of a loudspeaker's performance to balance the objective measurements done with test equipment. Often, a listening test will expose faults in a loudspeaker's performance that would not show up under standard objective testing. Publishing the results of listening tests, however, would sound like so much advertising copy: the response is "smooth", "transparent", "clean" and so on. Thus, the primary use of listening tests at a manufacturer is "in-house". That is, a manufacturer will listen to a loudspeaker to determine its subjective performance, then attempt to describe that performance in objective specifications (or to correct the performance if the listening tests are negative). For the user, a listening test before purchase is a good idea, especially if the loudspeaker is to be used for wide-range music. What is important in any listening test, whether by a manufacturer or a user, is that *all* conditions be rigidly controlled. For example, a listener will almost always pick the louder of two speakers as sounding better, all other conditions being the same.

AES and other standards organizations.

The AES Standards Committee, chaired by John Eargle of JBL, is currently working on a set of suggested loudspeaker measuring standards which will primarily cover techniques for specifying component loudspeakers. If adopted by the various manufacturers, this could greatly simplify comparisons between different manufacturer's devices, which is now a difficult task at best. Other standards organizations, such as the EIA, have done some work in this area, but they all have a similar set of challenges, namely the extreme complexity of the subject.

And . . .

You may have noticed that I have not presented much in the way of conclusions in this discussion. However, there are a few things that can be fairly stated as "true". First, the subject of loudspeaker specifications is very complex, and there are no easy answers to what should and should not be published, or what are reasonable standards. Second, whatever standards may be developed in the future should be developed for *loudspeakers*, and not be merely lifted from electronics specifications. In addition, the standards should indicate measurement *conditions* as well as *results* whenever possible and useful. Finally, if we as users continue to demand more detailed and accurate specifications from the manufacturers, we need to educate ourselves to be able to use what is supplied.



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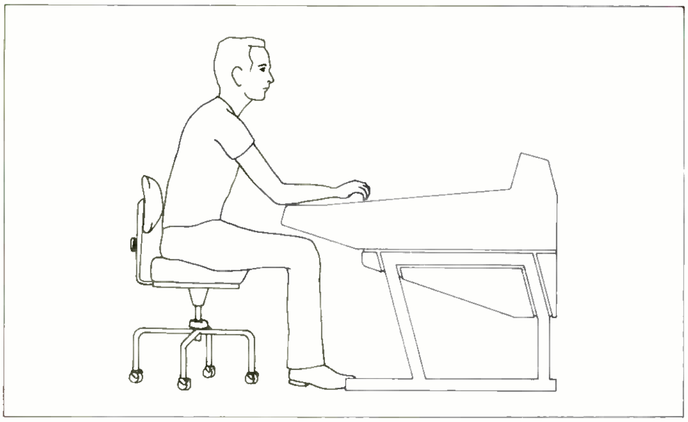

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What the

"Amplitude Transfer Characteristic"

can reveal about the differences in speakers—

by G.R. (Bob) Thurmond

Everyone who has listened to loudspeakers has observed that different models sound different. Everyone who has studied manufacturer's data has observed that many models have identical specifications. A little research will reveal different models which carry identical, or very similar, specifications, but which sound quite different.

Why? Are there sound qualities which we can hear but cannot measure, or are the published specifications simply inadequate? While the former alternative may always be true, many recent studies indicate that the latter is much more likely. In fact, it is often quite easy to measure performance differences between units which appear, on paper, to be alike.

Extensive measurements on home hi-fi type loudspeakers are widely performed these days, with the results published in national magazines for all to see. This is of little benefit to the audio professional, however, who uses custom systems, typically consisting of multiple horns and drivers, rather than package systems. Surely, someone has measured such professional components, but where are the results?

Most major manufacturers have conducted thorough tests on their components, as well as those of other manufacturers, and know pretty well how they perform. No doubt, many independent researchers have also run such tests. Yet, all these seem very reluctant to divulge even what tests they have run, let alone the results.

Some of this trepidation is understandable. Since there is no standard way to run such tests, everyone does it a little differently. Thus, trying to compare results from two different sources would likely present an apples-and-oranges problem. Furthermore, most manufacturers would rather not start throwing stones, as there is much glass in all their houses!

This author has run many hundreds of tests on professional loudspeaker components, and is quite willing to divulge the results. Over 100 models of horns and drivers, from all major manufacturers, plus several systems, have been tested. The samples come primarily from dealers and owners, with only a few directly from the manufacturers. In a number of cases, several samples, both new and used, were tested. Thus, the samples were average production units.

The actual tests are rather basic, consisting primarily of the on-axis frequency response. In addition, the input drive level and output sound pressure level are calibrated, so that on-axis sensitivity is simultaneously measured. Furthermore, the input impedance *vs.* frequency characteristic of drivers, and the response of horns at several angles off axis, are also measured. Together, these tests provide a fairly complete picture of the amplitude transfer characteristic, that is, how much output is produced by a given input at any frequency, for each device.

No attempt has been made to measure power-handling capacity, distortion, phase, or any of several other significant characteristics, for several reasons. In most cases, the necessary test equipment is not available. Furthermore, there are hundreds of devices, and thousands of horn-driver combinations, which are worthy of testing. To test all of these thoroughly would take almost forever. A better approach, it seems, would be to run a few basic tests on as many devices as possible, so this is exactly what is done.

Even so, many widely-used devices have not been tested. In many cases, this is simply because no sample has been made available. Another problem is that, for the most part, these tests are completely unfunded. Therefore, they can only be run as circumstances, such as work load, permit.

Even with all these limitations, a very useful body of data has been compiled. Admittedly, the amplitude transfer characteristic does not describe a device's performance completely, but it is probably the most important single factor. Furthermore, as already suggested, the purpose of these tests is not to "wring out" a device, but rather to allow objective comparisons between similar devices. In fact, the amplitude transfer characteristic for each device is distinctive enough to identify it clearly among other similar devices, and also can largely explain the overall "sound" of that device. Thus, these tests are quite sufficient for our purposes.

All tests are run in an anechoic chamber which has a working space 12 feet square and 7 feet high, which allows almost all measurements to be made at a distance of 8 feet from the device. At this distance, the sound field from all but the largest devices has "settled dowm" to its final nature. The chamber wedges are 3 feet deep, making the chamber quite anechoic above 100 Hz. and usable to 50 Hz. Below 50 Hz., however, results are not reliable.

The test equipment, and its configuration, is shown in Figures 1 and 2 for, respectively, the response and the input impedance tests. These procedures are commonly used, and the results obtained at other facilities have closely matched those obtained here. This speaks well for both the accuracy and the validity of the tests.

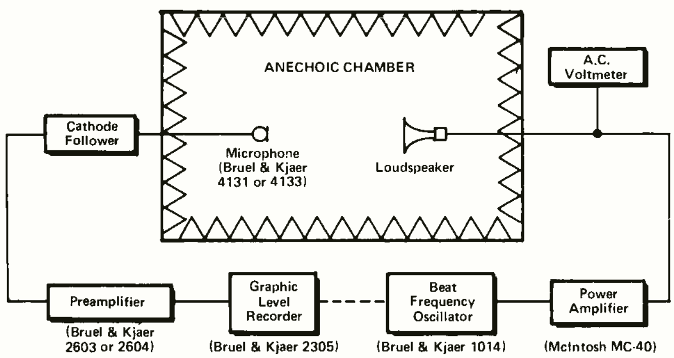

Figure 1: Equipment configuration for response tests.

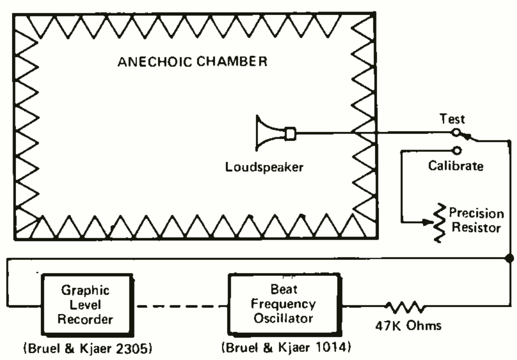

Figure 2: Equipment configuration for impedance tests.

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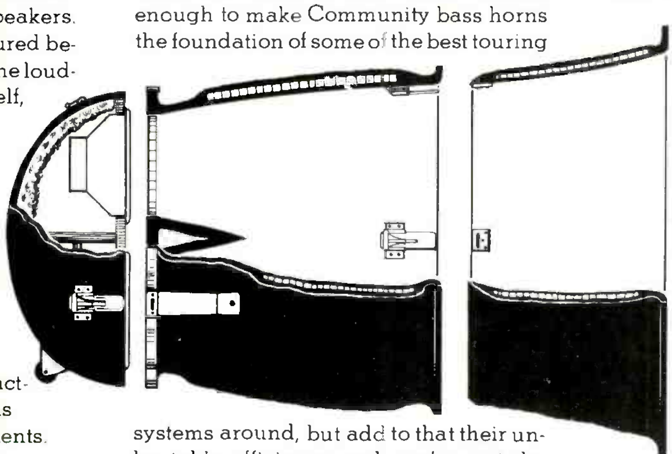

Not shown are our other bass horns: the FRC/B, designed to provide true horn performance in the smallest possible package, and the aptly named BLT, or Bass Long Throw, which does exactly that over several hundred yards with the closest attention to transients.

Like everything else that we make, our Levi, FRC/B and BLT are rock solid, portable, and built to last. That's reason enough to make Community bass horns the foundation of some of the best touring systems around, but add to that their unbeatable efficiency and you've got the bottom line for a full spectrum of professional applications.

What does efficiency mean? Because of our design criteria any Community bass horn's output is typically 4-6 dB above its wooden competitor's. To you, the professional sound person, this means that you need fewer bass horns to fulfill your requirements and, consequently, less drivers and electronics to power them. In addition, our bass horns weigh thirty to forty percent less than the old wooden horns meaning an additional savings in reduced installation and freight charges.

Need a couple of bass horns? See your Community dealer. You might only need one.

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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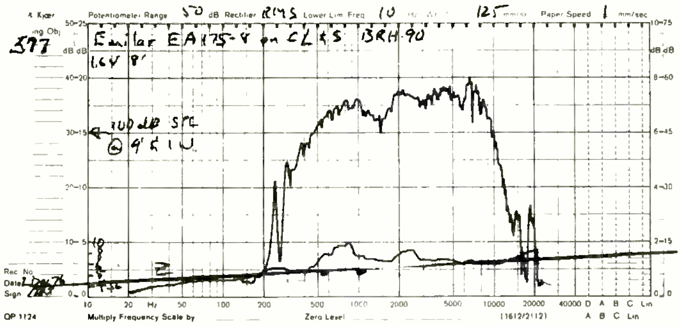
No matter how accurate and valid the tests, however, comparing results can still be very tricky. If the test results on two horn-driver combinations are compared, we cannot tell if the differences come from the drivers, the horns, or the particular combinations used. The only way around this dilemma is to reduce the number of variables. We can, for instance, test a number of drivers on the same horn and in the same way. Any traits which then appear on all such tests can be ascribed to the horn or the procedure, while a trait which appears on only one test is almost certainly caused by that particular driver.

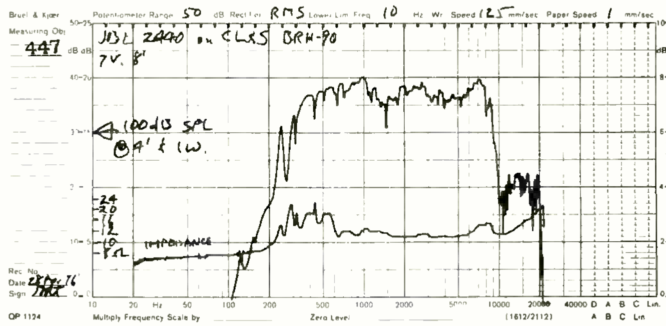
Thus we see that these tests can provide a valid means for comparing loudspeaker performance. It must be re-emphasized, however, that any test, taken by itself, has very limited significance. Only a comparison of related tests can yield really significant information.

As an example, let's look at the tests on a group of high-frequency drivers. These six drivers, from six different manufacturers, are all similar, but not identical, in design and specification. All have large Alnico magnets, an aluminum diaphragm, and a continuous power rating of 30 to 50 watts. All are widely used in high-level sound reinforcement applications.

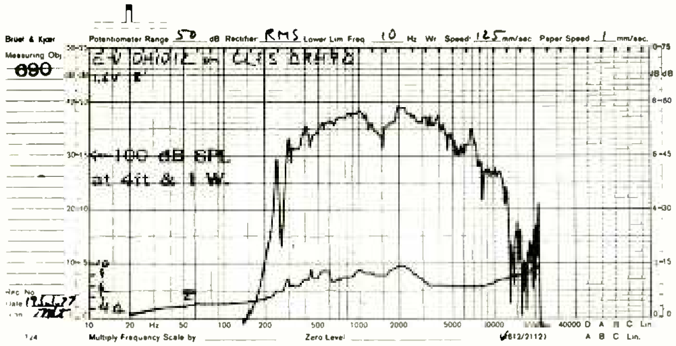
All drivers were tested on the same horn, a Community



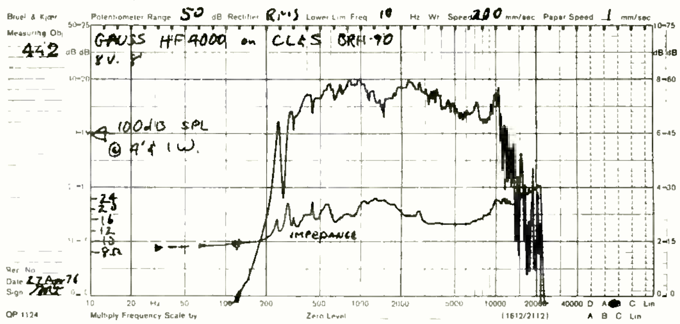
577 - EMILAR EA 175-8 on CL&S BRH-90



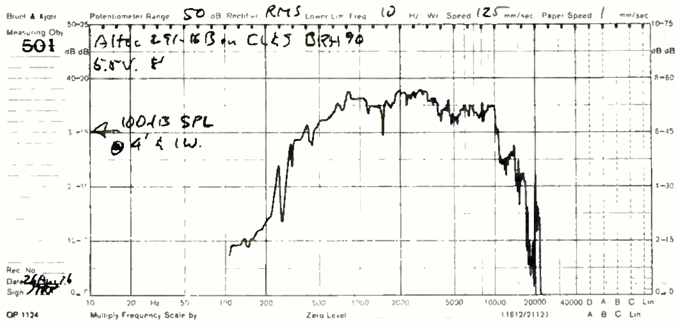
447 - JBL 2440 on CL&S BRH-90



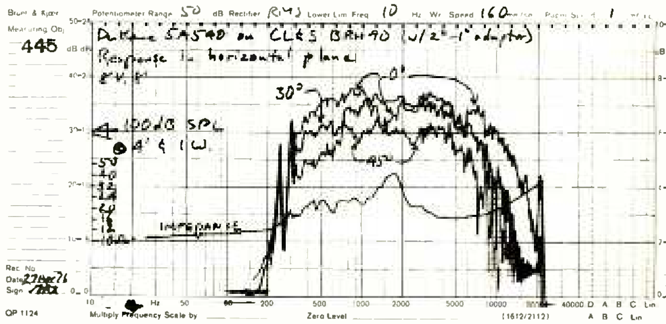
690 - E-V DH1012 on CL&s BRH-90



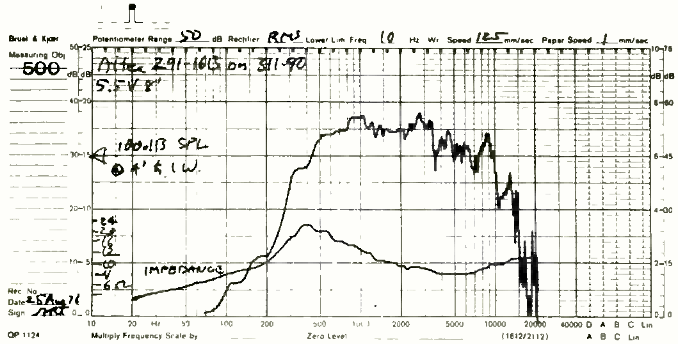
442 - GAUS HF-4000 on CL&S BRH-90



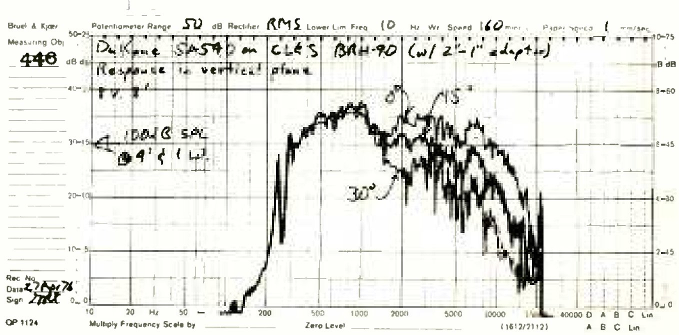
501 - ALTEC 291-16B on CL&S BRH-90



445 - DuKANE 5A540 on CL&S BRH-90 (Horizontal Plane)



500 - ALTEC 291-16B on 311-90



446 - DuKANE 5A540 on CL&S BRH-90 (Vertical Plane)

Light & Sound BRH-90, with appropriate throat adaptors. (Other tests have confirmed that a correctly designed and utilized adaptor will not degrade the performance of either the horn or the driver.) A perusal of these tests reveals several features which appear on all of them, and which, therefore, we can attribute to the horn. These include a broad dip between 1,000 and 2,000 Hz. plus several narrow resonance dips and one large one below 400 Hz. In addition, only the upper curve on the DuKane test should be considered, as the others were run off axis.

With these considerations in mind, a valid comparison between the various drivers can be made. We see that all have a response which is fairly flat, with differences at the frequency extremes. These differences may or may not be significant, depending on the particular application.

Differences in sensitivity are also evident. For this measurement, refer to the level calibration mark at the left side of each graph. That line is 100 dB sound pressure level at 4 feet from the mouth of the horn, with an input drive level which is equivalent to 1 watt. (More about this later.) All SPL readings must be made relative to this calibration mark. For instance, we see that the Gauss and JBL drivers produced 110 dB at 1,000 Hz. under these conditions, while the other drivers fell several dB below that level at that frequency.

Already you may have found some surprises in these graphs, but the best is yet to come. Consider, for instance, the input impedance curves. We see at once that the impedance varies considerably with frequency, and, for all but the Gauss driver, gets considerably lower than the rated value. This causes several complications, both in the testing and in the utilization of these drivers.

The impedance in the 500 to 1,000 Hz. range is important to the proper loading of a passive crossover network, while the lowest impedance determines the amplifier load and the power input for a given drive voltage. As evidence of the ambiguity this can cause, one manufacturer states that it rates one of its drivers at 16 ohms for professional use (because that is its impedance at crossover) and the same driver at 8 ohms for hi-fi use, (because that is its minimum impedance). For these tests, the minimum impedance is taken as the actual impedance, and the drive voltage set accordingly. It is felt that this is a realistic approach, but it results in some disagreement with the manufacturer's ratings.

The Altec 291-16B, for instance, is rated at 16 ohms. The measured value is close to this near crossover, but dips to 8 ohms at higher frequencies. With the drive voltage set, as it was, for a 8 ohm load, the measured sensitivity is lower than the manufacturer's rating. If the test were run with a drive voltage correct for a 16 ohm load, however, the sensitivity would come out at 3 dB higher, thereby easily meeting the manufacturer's rating. Again, we see the importance of knowing the exact test conditions.

It is probably already obvious, but in the case of the Altec driver, an impedance curve was not run on the Community horn. It was run instead when the drivers were tested on another, very similar, horn, so this is the curve shown. Experience has shown that impedance varies little between such similar horns, but may or may not be the same with different types of horns.

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- ☐ 69 Frazier F1290W (0°, 10°, 20°, 30° hor.)
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- ☐ 71 Frazier F10-2T S-N 62402 (tweeter max. and min., 0°, 45° hor.)
- ☐ 72 Frazier F10-2T (45° hor. and vert.)
- ☐ 74 Lafayette 6 in. No. 99C-01737 in 1 cu ft. ported box
- ☐ 75 JBL D-208 in std. ported box
- ☐ 76 Frazier F-1254, F4-110, Philips dome tweeter in Adkisson box
- ☐ 77 Frazier F1290W No. 2
- ☐ 78 Frazier F1290W No. 2 (45° hor.)
- ☐ 81 AR-4X (0°, 45°)
- ☐ 87 Frazier F5-2 Super Midget with beveled mtg. hole (0°, 45°)
- ☐ 88 Frazier F5-2 Super Midget with straight mtg. hole (0°, 45°)
- ☐ 89 Frazier 3½" cone tweeter (0°, 45°) No. 1
- ☐ 90 Frazier 3½" cone tweeter (0°, 45°) No 2
- ☐ 91 Frazier 3½" cone tweeter (0°, 45°) mtd. in Super Midget box
- ☐ 92 Frazier F5-2 with reflex ports blocked and open
- ☐ 93 Frazier F5-2 No. 3
- ☐ 94 Dukane 5A450 in standard box (0°, 45°)
- ☐ 95 Dukane 5A455 in standard box (0°, 45°)
- ☐ 96 Frazier F-333 (Atlas 2020) driver (unloaded)
- ☐ 97 Frazier F-333-590 (0°, 15°, 30°, 45° horizontal)
- ☐ 98 Frazier F-333-590 (0°, 15°, 30° vertical)
- ☐ 99 Frazier F1037W (0°, 45°)
- ☐ 100 2-Frazier F1037W stacked vertical (0°, 45° hor.)
- ☐ 101 Altec 290E (faulty) on 203B (0°, 15°, 30° hor., bad test)
- ☐ 102 Altec 290E (faulty) on 203B (0°, 15°, 30° ver., bad test)
- ☐ 103 Frazier F-1554 in std. box
- ☐ 104 Frazier F-1554 close to cone
- ☐ 105 Altec 421A close to cone
- ☐ 106 Altec 421A in standard box
- ☐ 107 JBL D130 close to cone
- ☐ 111 Frazier F5-2 Super Midget with modified ports
- ☐ 112 Frazier F5-2 Super Midget No. 2 with modified ports
- ☐ 113 Frazier F5-2 Super Midget close to cone and ports, normal and modified.
- ☐ 114 Frazier F5-2 Super Midget No. 2 close to cone and ports modified.
- ☐ 115 Frazier F5-2 Super Midget close to cone and ports, and at 4 inches, normal
- ☐ 116 Dukane 5A350 on JBL 2391 without lens (0°, 15°, 30°, 45° hor., bad test)
- ☐ 118 Dukane 5A425 in 6B425 box
- ☐ 119 Dukane 5A425 in 6B425 box close to cone and port
- ☐ 120 Dukane 5A350 on JBL 2391 (0°, 15°, 30°, 45° hor., faulty test)
- ☐ 121 Dukane 5A350 on JBL 2391 (0°, 15°, 30°, 45° ver., faulty test)
- ☐ 122 Dukane 5A405 on 5A420 (0°, 15°, 30°, 45° hor.)
- ☐ 123 Dukane 5A350 on JBL 2391 (0°, 30°, 60° hor.)
- ☐ 125 Frazier F5-2 S-N 52136
- ☐ 126 Frazier F-3X7
- ☐ 127 Dukane 5A350 on JBL 2391 with and without lens
- ☐ 128 Altec 808-8A, JBL 2410, and 2410 with phenolic diaphragm on Altec 811B
- ☐ 129 Altec 802D and 808-8A on 811B
- ☐ 130 JBL 2482 on 2356 (0°, 15° horizontal)
- ☐ 131 JBL 2482 on 2356 (15° vertical)
- ☐ 137 JBL 2410 with phenolic diaphragm on HL-180 (uncalibrated)
- ☐ 138 JBL 2410 with phenolic diaphragm and 2420 diaphragm on HL-180 without lens
- ☐ 154 Atlas 4040 on CL&S EC250
- ☐ 155 Matrecs 8 in. in std. box (uncalibrated)
- ☐ 156 Dukane 5A450 in std. box (uncalibrated)
- ☐ 157 Dukane 5A450 in std. box (uncalibrated, equalized flat)
- ☐ 158 Frazier F4-110 in small box
- ☐ 159 Frazier F4-110 in small box (equalized flat)
- ☐ 160 Frazier F4-110 in small box (equalized flat – second try)
- ☐ 161 Dukane 5A450 close to cone
- ☐ 268 JBL 2482 on CL&S RH60
- ☐ 269 JBL 2420 on Altec 811B (0°, 30°, 45° hor.)
- ☐ 270 JBL 2420 on Altec 811B (0°, 15°, 30° ver.)
- ☐ 271 Sunn Magna 136 on JBL 2355
- ☐ 272 Altec 808-8A on CL&S RH60 (0°, 15°, 30° hor.)
- ☐ 273 Altec 808-8A on CL&S RH60 (0°, 15°, 30° ver.)
- ☐ 274 CL&S LMF II with E-V 12L (aluminum(10°, 30°)
- ☐ 275 JBL 2440 on 2356
- ☐ 276 Sunn Magna 136 on JBL 2356
- ☐ 278 Altec 290E on 203B
- ☐ 280 CL&S MC12 with JBL K120 and E-V 1823 (0°, 30°, 45° horizontal)
- ☐ 281 CL&S MC12 with JBL K120 and E-V 1823 (0°, 15°, 30° vertical)
- ☐ 282 JBL 2482 on Altec 203B
- ☐ 283 JBL 2420 on CL&S RH60
- ☐ 285 JBL 2420 on CL&S RH90 (old)
- ☐ 286 JBL 2440 on CL&S RH60
- ☐ 288 JBL 2420 on Altec 311-90
- ☐ 289 Atlas PD-60 on Dukane 5A530
- ☐ 290 Sunn Magna 136 on Dukane 5A530 (0°, 30°, 45° hor.)
- ☐ 291 Sunn Magna 136 on Dukane 5A530 (0°, 15°, 30° ver.)
- ☐ 293 Dukane 5P355 on CL&S RH60
- ☐ 294 Sunn Magna 136 on Dukane 5A421 (0°, 15°, 30° hor.)
- ☐ 295 Sunn Magna 136 on Dukane 5A421 (0°, 15°, 30° ver.)
- ☐ 296 JBL 175 (0°, 15°, 30°)
- ☐ 297 JBL HL180F on CL&S RH60
- ☐ 298 JBL 2420 on Dukane 5A530
- ☐ 299 Sunn Magna 136 on JBL 2356 (0°, 15°, 30° ver.)
- ☐ 300 Sunn Magna 136 on JBL 2356 (0°, 15°, 30° hor.)
- ☐ 301 JBL 2440 on 2356
- ☐ 302 Atlas PD-60 on CL&S RH90
- ☐ 303 JBL 2440 on 2355 (0°, 15°, 30° hor.)
- ☐ 304 JBL 2440 on 2355 (0° 15° 30° ver.)
- ☐ 305 JBL HL180F on Dukane 5A421
- ☐ 307 JBL HL180F on Dukane 5A530
- ☐ 308 Dukane 5P355 on 5A421
- ☐ 309 Dukane 5P355 on 5A530
- ☐ 311 Dukane 5A540 on 5A421
- ☐ 312 Dukane 5A540 on 5A530
- ☐ 313 Altec 808-16 (aluminum) on CL&S RH60
- ☐ 314 Dukane 5A540 on CL&S RH60
- ☐ 315 E-V ST350 (0°, 45°, 60° horizontal)
- ☐ 316 E-V ST350 (0°, 30°, 45° vertical)
- ☐ 317 CL&S Leviathan with JBL 2220 (0°, 45° horizontal)
- ☐ 318 CL&S Leviathan with JBL 2220 with and without extension
- ☐ 319 CL&S Leviathan with JBL 2220 (drivers alone–horn removed)
- ☐ 320 Frazier F-3X7
- ☐ 323 Dukane 5A540 on JBL 2356
- ☐ 324 JBL 2405 (new style) (0°, 15°, 30°, 45°, 60° hor.)
- ☐ 325 JBL 2405 (new style) (0°, 15°, 30° ver.)
- ☐ 326 Dukane 5A540 on JBL 2355
- ☐ 328 JBL 2220 in E-V 606 box Z only (normal & step down)
- ☐ 329 JBL 2205 in E-V 606 box Z only (normal & step down)
- ☐ 336 JBL 2205 in E-V 606 box (normal and step down)
- ☐ 337 JBL 2220 in E-V 606 box (normal and step down)
- ☐ 338 JBL 2220 in E-V 606 box (1 2,3 and 4 units close together)
- ☐ 339 4 JBL 2220 in E-V 606 box in 2X2 stack (0°, 45°)
- ☐ 340 JBL 2482 on CL&S EC150 (0°, 20°, 30°, 40° toward curved sides)
- ☐ 341 JBL 2482 on CL&S EC150 (0°, 10°, 20°, 30° toward straight sides)
- ☐ 342 Same as No 340, horn mtd. in box (also 10°)
- ☐ 343 Same as No. 343, horn mtd. in box (also 10°) Z only
- ☐ 344 2 Super Midgets 8 ft. apart, measured 8 ft on axis of one
- ☐ 345 2 S-M 6 in. apart, measured 8 ft. away, between units and on axis of one unit
- ☐ 346 2 S-M 1 ft apart, measured 8 ft. away, between units and on axis of one unit
- ☐ 347 2 S-M 2 ft apart, measured 8 ft. away, between units and on axis of one unit
- ☐ 348 2 S-M 4 ft. apart, measured 8 ft. away, between units and on axis of one unit
- ☐ 349 2 S-M 8 ft. apart, measured 8 ft. away, between units and on axis of one unit
- ☐ 350 Dukane 5A455 with 70V. transformer @ ½ watt Z only
- ☐ 351 Dukane 5A455 with 70V. transformer @ 2 watts Z only
- ☐ 352 Dukane 5A455 Z only
- ☐ 353 Frazier F5-2 (modified) Z only
- ☐ 354 Atlas PD-60 on CL&S RH90 Z only
- ☐ 355 JBL D130 in standard box Z only.
- ☐ 357 RCF D3045 on CL&S RH90 Z only
- ☐ 358 RCF 4060 on CL&S RH90 Z only
- ☐ 360 2 JBL K140 on CL&S Standard Leviathan with Z
- ☐ 361 2 JBL K140 in CL&S Standard Leviathan with Z (drivers only)
- ☐ 362 RCF 4060 on CL&S RH90
- ☐ 363 RCF D3045 on CL&S RH90
- ☐ 367 1 JBL K140 and K145 in Jensen with box with Z
- ☐ 368 2 JBL 2220 in Jensen with box (back off and on) with Z
- ☐ 369 JBL 2420 (with 4 mfd cap) on CL&S LRH with Z
- ☐ 370 3 JBL 2420 on 3 CL&S LRH in boxes, side by side
- ☐ 371 JBL 2220 in Altec 825 with Z
- ☐ 372 4 wide array of JBL 2420 on CL&S LRH in boxes (0°, 30° horizontal)
- ☐ 373 Emilar EA175-8 on CL&S RH90 (old) with Z
- ☐ 374 JBL K140 in standard box with Z
- ☐ 375 Altec 604E in closed box with Z (0°, 30° horizontal)
- ☐ 376 2 Altec 421A (seriesed) in 210 with Z
- ☐ 377 2 JBL 2220 in Altec 210 with Z
- ☐ 378 Altec 421A in standard box with Z
- ☐ 380 2 JBL 2420 on 2 2309 with cap. in box (B.S.) with Z (0°, 15°, 30°)

Listings continue on page 87 . . .

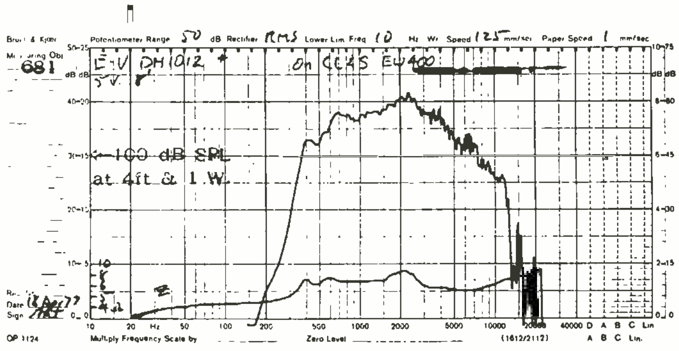
A couple of incidental comparisons are worthy of note. Graphs Nos. 500 and 501 show the same driver on two different, but very similar, horns, thus allowing a comparison between the horns. The Altec horn appears to have a smoother low-frequency response, while the Community horn is better at the high end. These differences, however, are not nearly so great as those between the various drivers.

All these drivers were new when tested. What happens to their response after long usage? For that matter, how consistant is the performance from one unit to another? We have at least some partial answers to these questions.

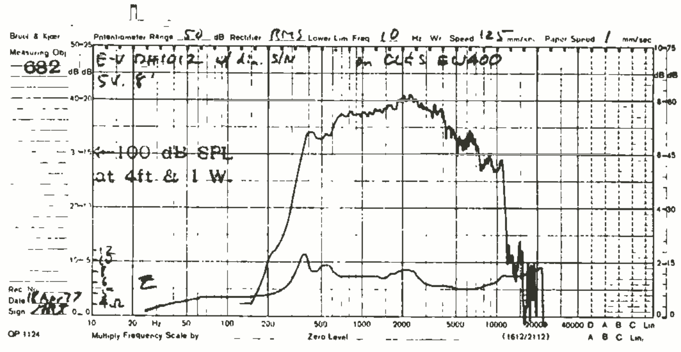
Graphs No. 681, 682, and 683 show three different E-V DH1012 drivers on the same horn (but not the same as before). Test No. 682 was on a well-used driver, while the other two were new. The only difference seems to be at the highest frequencies, and even that is slight.

Tests Nos. 660 and 661 were run on two used JBL 2440's. Comparing these with Graph No. 447, we again see that the only difference is a slight change at very high frequencies. This consistency speaks well not only for the drivers, but also for the test procedures.

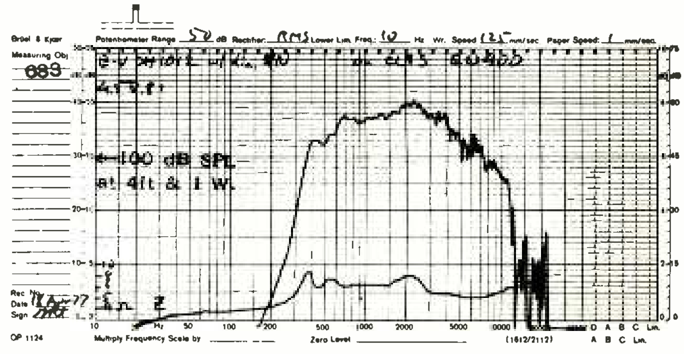
Graphs Nos. 625 and 596 show several samples of new Dukane and Emilar drivers, respectively, all plotted on the same



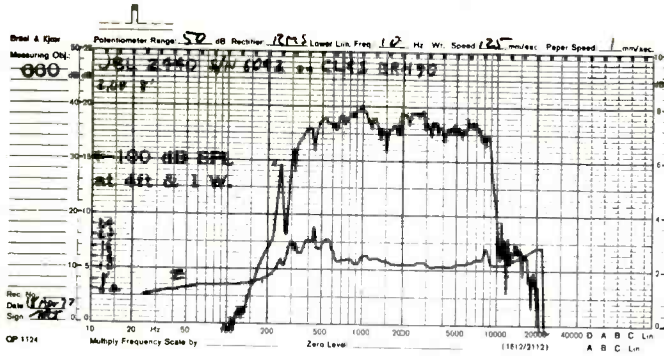
681 - E-V DH1012 on CL&S EW400



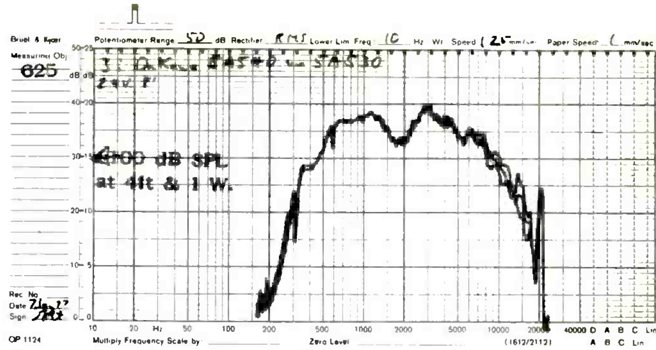
682 - E-V DH1012 (used) on CL&S EW400



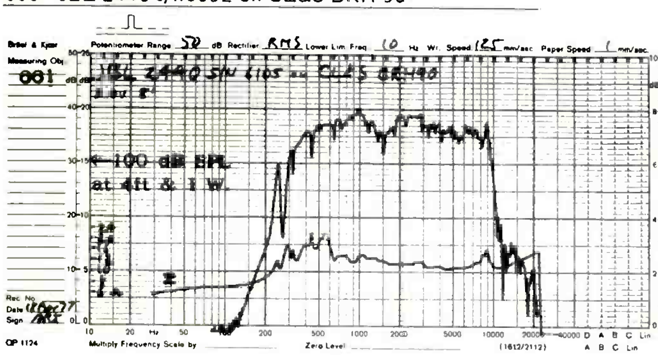
683 - E-V DH1012 (different) on CL&S EW400



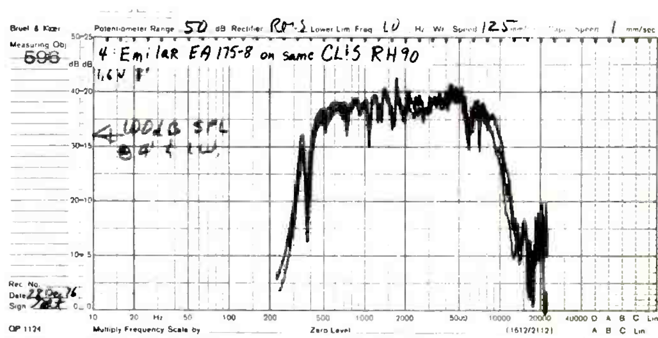
660 - JBL 2440 s/n6092 on CL&S BRH-90



625 - 3 DuKANE 5A540 on 5A530



661 - JBL 2440 s/n6105 on CL&S BRH-90



596 - 4 EMILAR EA175-8 on same CL&S RH90

paper. Again, we find the unit-to-unit consistency to be very good, with some variations at very high frequencies. Please note that these tests were run on horns which also are different from the others, thereby making it difficult to compare these curves with others.

We have seen that even basic tests such as these can easily pick out the differences, as well as the similarities, between similar models of horns and drivers. We have seen that it is possible to isolate the unit which is responsible for certain characteristics, and to obtain very consistent results. Many other such comparisons are possible, and we shall explore some of them in the future.

SYSTEM V ...

a realtime/software/minicomputer approach to digitally processed audio

by Daniel Coren and Harry Mendell

Digital systems, as they are most generally applied in the audio industry today , are a relatively inflexible design. A flanger flanges, a delay line delays, a pitch changer changes pitch, all within the limits of the design of the respective device. Those familiar with the use of digital audio devices are aware that function does not extend very far beyond the nomenclature of the system of interest.

Powerful as the common rigidly-configured digital audio signal processors are, the advent of the miniature digital central processing unit and low-cost memories of recent years have radically changed the scope of practical possibilities open to the audio signal processing system designer. The fruit of these advances in digital system technology will begin to percolate down to the user level in the very near future. The impact of audio processing systems that can be made to perform a variety of audio signal transformations with no changes in the configuration of the equipment itself can be mind boggling even to the sophisticated audio-digital technologist. Imagine, for example, being able to obtain delay, reverberation, and pitch-change from a single device with only a command from a keyboard. This very possibility is the subject of this article.

The equipment that makes these capabilities possible is minicomputer-based digital systems employing a repertoire of signal processing programs that enable user modification of the program being accessed at any given time. The hardware/software system is called the System V and is available in a price range that would make it attractive for commercial application by recording studios and by live performers.

About the Authors . . .

DANIEL COREN holds an M.A. and Ph.D. in music from the University of California. He was the director of the Presser Electronic Music Studio at the University of Pennsylvania for seven years and has worked on the applications of computers to sound for the past three years.

HARRY MENDELL holds an M.S. degree from the Moore School of Electrical Engineering at the University of Pennsylvania. His masters thesis was a research project for facilitating the design of an artificial eye for the blind. The project involved the design and implementation of a computer controlled camera system for improving the image a blind person would receive from an implantable artificial eye.

The primary feature that makes the System V innovative and particularly noteworthy is the fact that it is capable of performing its functions in "real time", that is, at the time and rate at which the signal to be processed occurs. The interactive nature of the available System V programs make use of the system an easily-learned skill for those who are not technically inclined.

Since the efficiency of System V's software allows it to run completely in real time, the system is ideally suited for live performance. As a performance instrument, its impact is truly staggering; for the first time, musicians are able to take the sophisticated studio effects heard on their recordings directly into the concert hall in one compact, easy-to-manage package. Effects that might take many hours of painstaking work with

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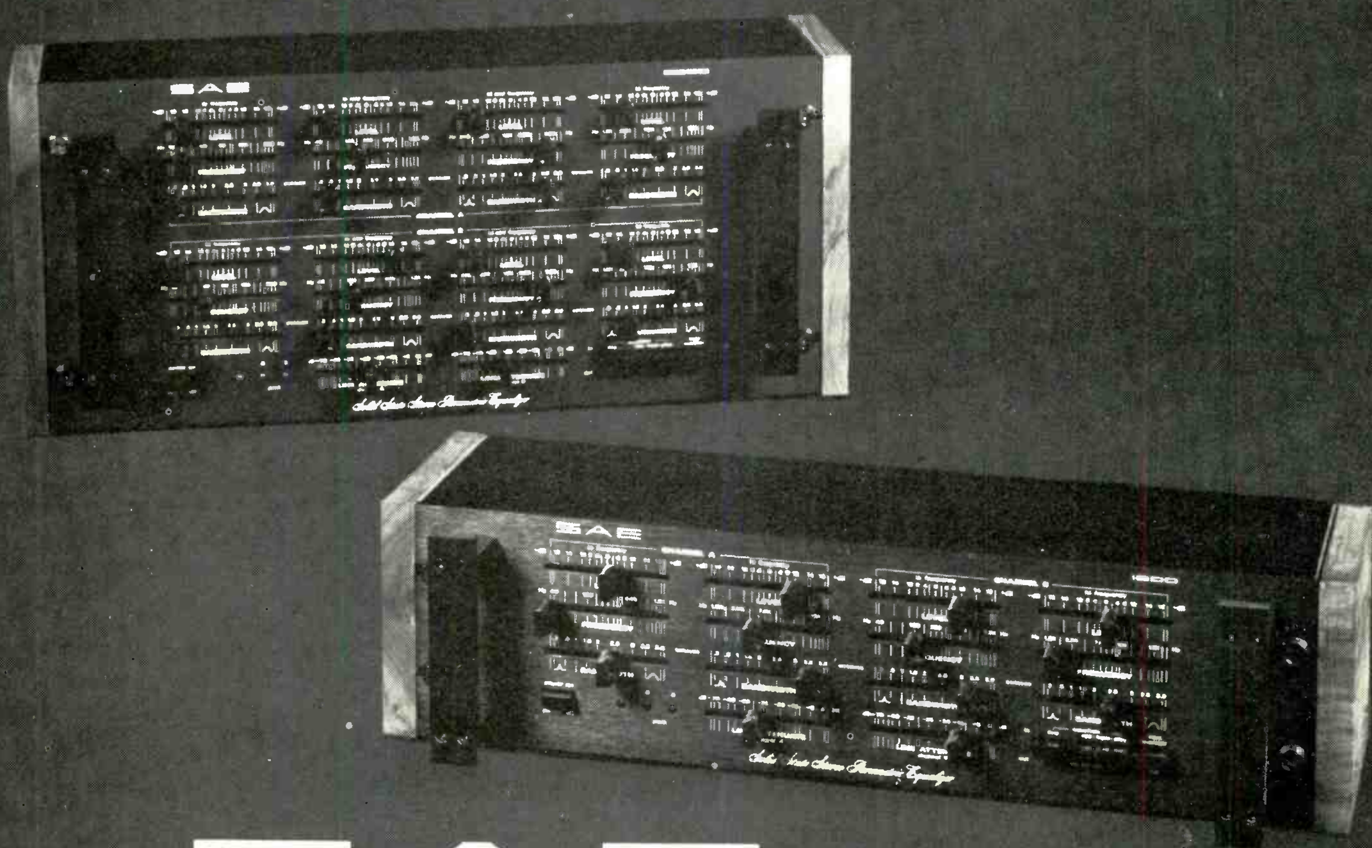

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traditional studio equipment can be preprogrammed and recalled with a few keystroke commands from the system's video terminal keyboard. Or, during live performance, the system can be played from its keyboard in an intuitive, improvisational way.

For example, a musician playing from the system's keyboard could make it seem that a concert hall is continuously changing its dimensions. Or, again by playing the system keyboard, a musician can create layer upon layer of sound, the layers blending into each other and slowly fading away – an effect that is today completely unavailable in live performance.

System V also has the potential of revolutionizing the audio technology of the movie industry. In a science-fiction film, System V could do for sound what Kubrick's *2001* did for visual imagery; the soundtrack of any movie made with System V would gain richness and subtlety unprecedented in the cinema.

Before discussing the operating capabilities of System V, a discussion of some of the technical considerations common to all digital sound processing systems may be helpful.

The fundamental steps that must take place in a computer-based sound manipulation system are as follows:

1. Sound, which exists as fluctuating pressure in the atmosphere, is transduced to electrical form, that is, is translated from fluctuating pressure to fluctuating voltage.

2. Once it is in the form of a fluctuating voltage, the signal is *digitized;* that is, it is changed from a continuous function to a series of discrete numbers. The device that digitizes an electrical signal is called an analog-to-digital converter, or ADC. Each number the ADC creates is called a *sample;* the speed at which the converter creates these numbers is its *sampling rate*.

3. In digitized form, the signal can now be stored in a computer's memory and manipulated by computer programs.

4. When the computer is done processing the digitized signal, the numbers in the computer's memory are changed back to electrical form by a digital-to-analog converter, or DAC.

5. Finally, the electrical signal is changed back to a pressure function in the atmosphere, where it reaches the ear as sound.

There are four types of distortion that can arise in digital sound systems: *quantizing noise, sampling rate noise, aliasing,* and *DAC "glitches."* In order to understand these problems, let us consider a situation in which the program in the computer does nothing more than accept numbers from the ADC and pass them out immediately to the DAC.

How closely will the output of the system match the original input? In general, a comparison of the output of the DAC with the input to the ADC will look like the trace that appears in Figure 1.

While the input to the system was continuous, the output is a series of discrete steps, these steps being the individual samples. This is the most distinctive feature of digital signal processing, and the problems mentioned above are all aspects of this phenomenon.

How can these problems be alleviated? First of all, we want the numbers supplied to the computer by the ADC to be as accurate a representation as possible of the fluctuating voltage input. It is clear that the more numbers the converter can represent, the better off we will be. Since the original voltage input to the ADC is a continuous function, it would take an infinite number of discrete steps to represent it absolutely accurately, since there are an infinite number of points along any curve. Therfore, any digital representation of an analog waveform will be prone to this particaular type of inaccuracy, which is called quantizing noise. For high quality audio signal processing, a converter that can represent a few thousand different numbers is necessary. Computer Music's ADC can represent 4,096 number; that is, it can represent numbers up to 12 binary digits long, and is therefore called a 12 bit converter.

Although bits per sample is the most often mentioned specification of ADC's, the number of bits a converter produces represents only the theoretical upper limit of its performance. In practice, many converters introduce much more noise than one would expect from their specified number of bits. Many converters that are perfectly suitable for the measurement of low frequency signals such as control voltages, strain gauges, tec., are unsuitable for audio applications. For example, a converter might make an erroneous conversion once every hundred samples. In low frequency applications, a computer's averaging program could easily smooth out such an error, but in an audio application, such an error rate would give rise to a distinctive gravelly sound or a high whistle. The selection of an ADC must be a process of trial and error; this task is made more difficult by the fact that ADC specifications are often very sketchy.

Just as important a consideration for an ADC is its sampling rate. It is obvious that if the converter could only create one number per second from its input

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voltage, it would be of little use, no matter how many different numbers it could represent. The same could be said of a movie camera with film that can represent a wide range of shadings but which could only take pictures at the rate of one frame per week.

It can be shown that in order to sample a waveform of a given frequency accurately, the sampling rate must be at least twice that of the given frequency. Practically speaking, it should be three times as fast or even faster.*

Since the ear can hear frequencies up to about 20 kHz, an ideal (but impractical) sampling rate for a digital sound system would be in the range of 40 to 50 thousand samples per second; 25 thousand samples per second is usually sufficient for practical purposes.

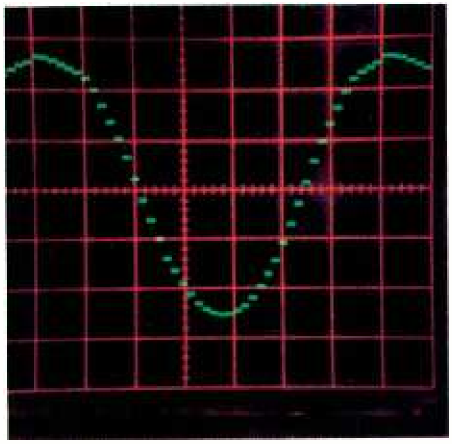

Figure 1: DAC output taken ahead of the low-pass smoothing filter. The discrete steps represent the analog values of the instantaneous numerical sample values appearing sequentially at the DAC input.

Sample Rate Noise

Let us assume that our ADC has sufficient precision and runs sufficiently fast. The output of the DAC is still discontinuous, and even if the fundamental frequency of the output is the same as the fundamental frequency of the input, the discontinuities in the DAC output will introduce high frequency components that were not present in the original signal. These unwanted high frequencies are sampling rate noise. One way to eliminate sampling rate noise is to maintain a sampling rate high enough so that the noise will be above the range of hearing. Since this solution is not practical, especially if the system is to run in real time, it is necessary to put the signal from the DAC through a smoothing low-pass filter.

Aliasing

Suppose the input to the ADC contains frequencues above the audio range, much too high to be sampled accurately. In this case, the sampling process itself can introduce unwanted frequencies that were not present in the original input as illustrated in the following diagram. (Figure 2)

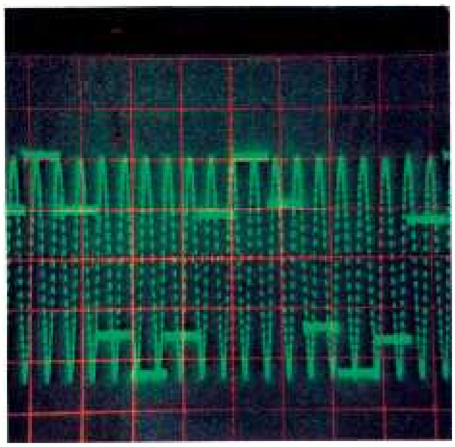

Figure 2: The phenomenon of "aliasing." The DAC output resulting from conversion of a sine wave of approximately twice the sampling rate. The DAC output is superimposed over the trace of the sine wave input signal.

This phenomenon is called aliasing. These aliasing frequencies, which can be far lower than the input frequencies that produced them, cannot be eliminated by the smoothing filter after the DAC; therefore, high frequency components that might give rise to aliasing must be eliminated by another low-pass filter before the ADC.

Glitches

DAC "glitches" arise because DAC's do not behave with theoretical perfection. Theoretically, when a DAC switches from one sample to the next, a rapid and smooth change in output voltage will occur. However, because of the differing switching times of current or voltage switches in the DAC, large voltage spikes can be created in the output.

This situation is aggravated when many bits change from sample to sample, since many switches will then be changing state. The solution to the "glitch" problem is to use a sample and hold that is timed to sample the output of the DAC after it has settled to its new value and to hold that sample until the next conversion has settled.

The reader may conclude from the brief discussion that digital systems are complicated entities. They are also delicate ones and subject to disasterous disruptions of their operation from interactions with their environments. All digital systems that are required to interact with systems outside their boundaries require interfacing. In the case of the

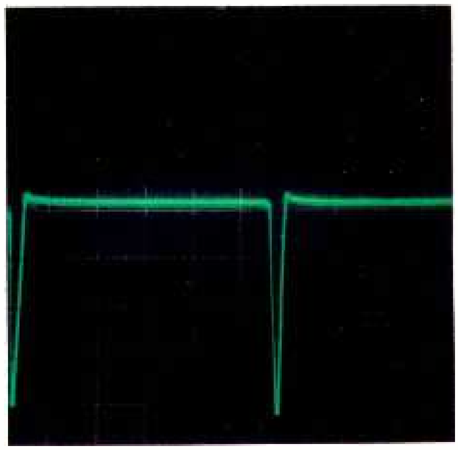

Figure 3: DAC Glitches. A DAC output resulting from the conversion of alternating binary numbers 011111111111 and 10000 0000000. These binary numbers represent decimal values of 2047 and 2048, respectively. The percentage change in numerical value is less than 0.05%. The result of differing switching times is clearly illustrated.

System V, the operator-machine interface is provided by the keyboard and the video terminal. It is also necessary to provide interfacing between the digital system and the audio systems with which it is to be used.

Analog Interface

The analog interface to the computer system consists of an actively balanced input having a 600 ohm resistive load impedance. The input buffer then feeds a scaling circuit that permits adjustment of the incoming signal to a range that will result in maximum utilization of the conversion range of the ADC. The ADC has a maximum digitizing range of 5 volts peak. Input signal excursions beyond this range are clipped by the full-scale capability of the ADC before clipping is reached in the input buffer and anti-aliasing portions of the interface. The available input headroom is thus adjustable by the user.

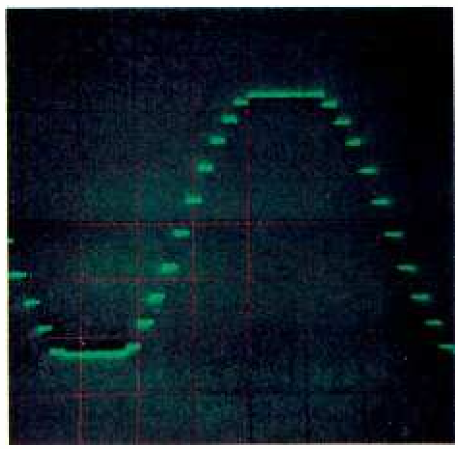

Figure 4: The clipping characteristic of the ADC/DAC system as seen at the output of the DAC. The clipped levels are represented by the binary numbers 111111111111 and 000000000000. The asymmetry of the signal is due to characteristics of the signal source, a Moog synthesizer.

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* *For a rigorous discussion of sampling rate and quantizing noise, see Max V. Mathews' "The Technology of Computer Music".*

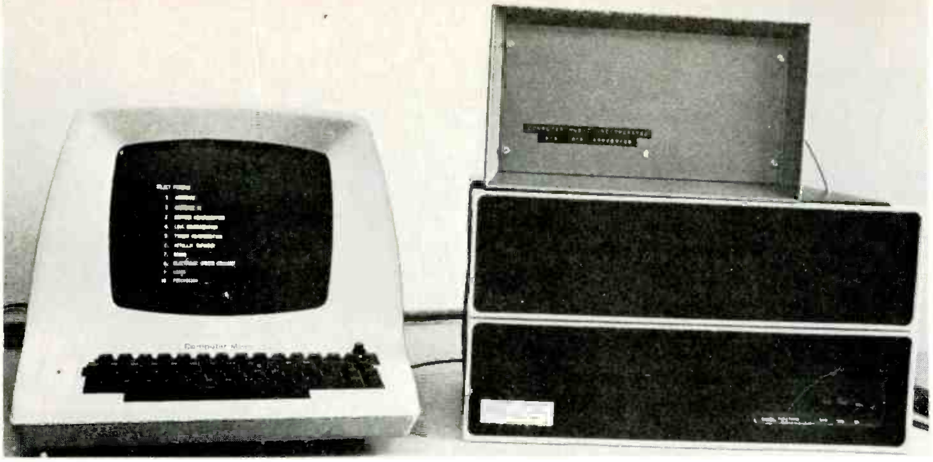


Figure 5: The COMPUTER MUSIC "SYSTEM V".
The System V consists of the video terminal, the DEC PDP-8 computer, and the interface unit. The video terminal may be located remotely from the other parts of the system.

The output buffer circuit is arranged in an actively balanced configuration permitting a 600 ohm load to be driven over a range of signal levels. Output signal clipping occurs at above +20 dBm, provided the input/output buffers are adjusted so that digital clipping does not occur first.

Once the problems discussed here have been solved, a system that exploits the benefits of digital sound processing can be created. System V is such a system.

No knowledge of computer programming is necessary to operate System V. The system's software is controlled entirely by keystroke commands from a video terminal and is designed for fast, intuitive interaction with its users.

Perhaps this is an excellent point at which to describe the system at hand and its abilities.

In the recording studio, System V can create sound and implement techniques that producers and engineers could only dream about before. In order to duplicate the capabilities of System V, an engineer would need a large array of existing studio equipment — a tape machine with adjustable heads and variable tape speed, a digital delay system, echo chambers, reverberation plates, and a synthesizer — and even then he would lack the quality and versatility of a digital system.

For example, taping methods used in studios today often require several recorders, custom playback head assemblies, and good luck. Even with a simple technique like tape echo, the audio signal becomes distorted very quickly and is covered by tape hiss after only a few echoes.

System V can simulate a tape recorder with multiple playback heads and variable tape speed that is controlled entirely from the video terminal keyboard. This simulated tape machine can create tape echo that has no appreciable noise build-up and very low distortion. This tape echo, which has always been limited by technical problems, now at last becomes a powerful, flexible music tool, offering a studio engineer possibilities that he may have wished for but was unable to achieve. These new possibilities are typical of the effect real time digital processing, high quality signal, and ease of operation will have on traditional studio techniques. All this can be achieved at prices ranging from \$15,000 to \$21,500, depending on equipment selection.

The System V price list is as follows:

System V/8 — PDP 8A400BM computer, 8K core memory, \$15,000.

—Computer Music SDRS-100 Sound Digitizer and Reconversion system with two outputs.

—Lear-Siegler ADM-100 computer video terminal.

—System V RTSP-5 Real Time Sound Processing Software.

—Installation and 90-day warranty.

System V itself is a minicomputer-based system for the real time processing of audio signals. It is capable of creating new and exciting sound effects that are available by no other means; it can also be used as a tool for simulating a great variety of acoustical environments. System V can intensify and enrich the sound of specific instruments; capture and manipulate sound; enhance the capabilities of any studio; be controlled at the push of a button; and remember and recall settings such as head placement.

The system is equally well suited for live performance or for studio work. New features are easily added to System V as they become available.

Until now, the available methods for modifying audio signals — tape machine, reverberation plates and the like — have been limited either by inflexibility, distortion, or both. Real time digital processing now eliminates these problems because it is computer-based, and can therefore manipulate its audio input while the signal is stored in digital form in the computer's memory. As a result, the output of the system is free of the distortion characteristic of mechanical devices such as reverberation plates or spring reverberation systems. Since System V simulates the acoustics of reverberant spaces by implementing a proprietary mathematical model, its output does not contain the noise introduced by standard feedback techniques like tape echo in which the analog signal is fed back again and again through an electronic system. Furthermore, the most sophisticated state-of-the-art techniques have been used in interfacing the analog to digital and digital to analog converters with the system's computer, so that the noise problems typical of a/d and d/a conversion have been virtually eliminated.

One of the most innovative aspects of System V is the efficiency of its software. Until now, computer simulations of acoustical environments have involved completely dedicated large computers (such as the PDP-10) which have operated roughly 1/30th as fast as real time. The design of System V's software allows an industry standard minicomputer, the PDP-8, to run entirely in real time while still allowing a frequency response typically between 10 and 15 kHz.

Software

The software of the system is divided into three parts, Tape Techniques,™ Soundspace,™ and Pitchshift.™ With Tape Techniques, System V can simulate a multiple-head, variable speed tape recorder, a recorder that can be controlled simply and with unprecedented precision, flexibility, and performance.

Soundspace offers a choice among 10 acoustic environments. These environments differ in their sound absobtion characteristics (that is, their "liveness"), their size, and their decay times. Unlike echo chambers and reverberation plates, whose reverberation characteristics are inflexible and that reverberate all inputs in the same way, each of the programs in Soundspace is optimized for a particular kind of sound.

Pitchshift is new software currently under development which will be offered as part of the System V package at the time of delivery to its users. Pitchshift accepts small segments of audio input and plays them over and over while modifying their original pitch in various ways. With Pitchshift, it is possible to obtain from a single input tone such effects as continuously repeating glissandi in either direction, repeating sequences of tones, and transposition by a constant interval.

Here is an example of how Tape Techniques can alleviate the problems of standard studio techniques. Suppose a recording engineer wanted to arrange four playback heads on a multi-head recorder so that a sound would be repeated in a particular rhythmic pattern, feeding one of the playback heads to the record head, thus creating a complicated pattern of delays and tape echo. System V can simulate this situation in a matter of seconds. The positions of each head as well as the amount of tape echo can be instantaneously and precisely controlled from the system's video terminal keyboard.

Such a configuration could be achieved with a four-head tape recorder, perhaps in conjunction with a digital delay unit, but only with a great deal of effort. Correctly adjusting actual playback heads is a tedious, time-consuming task, and even after the adjustment is made, the tape hiss and distortion problems of actual tape echo and of delay lines would still be there to contend with.

System V can be programmed to store up to 20 simulated tape delay configurations; any configuration is immediately accessible from the video terminal keyboard. While the system is operating, fine adjustments in head placement can be made from the keyboard; in fact, the video terminal keyboard really functions like the keyboard of a musical instrument.

System V can instantaneously form simulated tape loops at a single keystroke command from the terminal keyboard and these loops can be varied in speed over a 10 to 1 range. A word or series of notes can be captured and moved up and down in tempo and pitch – moved down to create drawn-out thunder-like effects, moved up until they become rapid blurs of sound.

No tape machine can create a tape loop on the spot, and no tape machine has the variable speed capabilities of System V. After an actual tape loop has been spliced, it will almost always contain an objectionable click at the point of the splices; even digital delay units capable of creating loops instantly cannot remove these clicks. System V's simulated tape loops are created by Smoothsplice,™ a special program that produces click-free splices.

Each of the 10 different Soundspace programs produces a distinctive reverberant effect, and each is optimized for a particular kind of sound. Furthermore, the entire reverberant structure of some of the Soundspace programs can be altered from the video terminal keyboard. For example, in one Soundspace program (called Timbre Reverberation) the keyboard can be played during the execution of the program to dynamically change the sound of the input. Each key is assigned different acoustic characteristics, and therefore each produces its own distinctive kind of coloration. This kind of control of sound cannot be produced even with the most sophisticated synthesizers and filters.

Also, since stereo and quad outputs are available from System V, spatial depth and perspective become a controllable aspect of the reverberant Soundspace programs. From the video terminal keyboard, an engineer can put his audience's ears inside a set of drums or a mile away.

The Pitchshift programs simulate a tape machine that can record a sound on a tape loop and immediately play the loop back at varying speeds. After the loop is played a predetermined number of times – from once to over 4,000 times – the program automatically switches into a trigger state, waiting to record and play the next sound it receives. Pitchshift can be used for a variety of exotic effects: it can create glissandi from piano chords; it can play guitar notes over the entire audio range; it can repeat a single input tone in repeating melodic patterns like a synthesizer sequence; it can transpose a single input tone by a constant interval.

Pitchshift switches from record to play mode and back automatically and is suitable for live performance we well as for studio work. The transposed sounds it produces are distortion free, unlike the output of a harmonizer, and a harmonizer cannot even begin to compare with the frequency range of Pitchshift. Needless to say, no tape transport or tape recorder electronics have these capabilities either.

Adding to the powerful flexibility of the signal processing portion of the System V software package, an interactive prompting response has been devised to assist the user in selecting the software functions he requires. As can be seen from the following example, very little experience is required for system utilization.

When System V is turned on, a message appears on the screen of the video terminal:

SELECT EITHER
1. TAPE TECHNIQUES
2. SOUNDSPACE
or

3. PITCHSHIFT

The user responds by typing either 1, 2, or 3.

Let us assume that Tape Techniques is selected. In this case, the system responds with a series of requests for information about the tape machine the user wishes to work with.

First, the system asks:

1, 2, 3 or 4 PLAYBACK HEADS?

The user responds by typing in the appropriate number. The system then responds:

SELECT POSITION OF
PLAYBACK HEADS

The user is now able to specify the relative positions of the playback heads he has selected by typing the number of the particular playback head followed by a number from 1 to 80. The higher the placement number, the further the simulated playback head will be from the record head.

As the positions of the heads are specified, their relative placements are displayed on the video terminal screen. For example, if the user selects 4 playback heads, and positions them by typing 1:20; 2:40; 3:60; 4:80; the placements of the heads on the screen will be:

1 2 3 4

If 2 playback heads are selected, with the specifications 1:40; 2:41; the display of their positions will be:

12

This last example demonstrates that a digitally simulated tape machine can go beyond the physical limitations of an actual one. If the heads are placed this close together, the time interval between them will be about 1 millisecond. If an actual tape machine were running at 15 ips, it would have to have heads small enough to be placed .015 inches apart to achieve such a short time delay.

The playback heads do not have to be placed in numerical order, for example:

2 3 4 1

is a possible configuration.

After positioning the playback heads, the system asks:

FEEDBACK FACTOR FOR
PLAYBACK HEAD 1?

to which the user responds by typing a number from 0 to 7. By typing this number, the user selects what fraction (from 0 to 7 eighths) of the output from playback head 1 will be fed back to the simulated record head, or, in other words, he controls how much tape echo will be used. (Playback heads 2, 3 and 4, if they are selected, do not function as feedback heads.)

The range of tape echo effects offered by System V far surpasses what can be done with an actual tape recorder. If playback head 1 is placed at the maximum distance from the record head (that is, if its placement specification is 1:80) and if a feedback factor of 7 is selected, a series of echoes can be made to last for 15 seconds before decaying to silence. If such a situation were implemented on an actual tape machine, the echoes would quickly become degraded in quality by tape hiss. The tape echo produced by System V, because it is created digitally, will remain clean until it has entirely decayed.

If smaller feedback factors are selected with closer placements of the feedback and record heads, subtle echo effects unavailable on conventional tape machines can be achieved. If a feedback factor of 0 is selected, Tape Techniques becomes, in effect, a digital delay system with from 1 to 4 outputs.

After the feedback factor has been selected, the system asks:

VARIABLE SPEED?

to which the user responds by typing Y for yes and N for no. By selecting this option, the user gains the ability to simulate the dynamic control of tape speed while the program is operating. With this option, the time intervals between the record and playback heads can be varied (although the relative distances between the heads will not be changed). Variable speed increases the decay time of simulated tape echo to as long as a full minute.

After the user's response to the variable speed option, Tape Techniques automatically begins excution. The outputs of the simulated playback heads are digitally mixed by the computer and are fed to one or two output channels at the user's discretion. (In the quadraphonic version of System V, the output of each playback head is fed to its own channel.) The placements of the playback heads and the feedback factor for playback head 1 can be altered while Tape Techniques is running. The heads are positioned just as they were during initialization. For example, typing in 2:56 during execution will position playback head 2. A new feedback factor can be obtained by striking the key labelled "feedback" on the video terminal keyboard and then typing in the appropriate number.

Suppose that the original configuration consisted of 2 playback heads, the first positioned at 10 and the second positioned at 40, and that the feedback factor was 4. If during execution the user desires more time between the playback heads and a greater feedback factor, he can type in 2:70 to move the second playback head further from the first and then "feedback" followed by 6 to increase the feedback factor from 4/8 to 6/8.

During execution, the lower three rows of the video terminal keyboard can be used to dynamically control the placement of playback head 1 and (if the variable speed option was selected) the simulated tape speed. Upper case characters (that is, characters struck while the shift key is depressed) control simulated tape speed and lower case characters control the placement of playback head 1.

What would normally be the space bar on the video terminal keyboard is labeled FREEZE™ on the System V terminal. If it is struck while the program is executing, a segment of audio input will be played repeatedly, thus simulating a tape loop. The simulated loop can be from a tenth of a second to several seconds long, depending on the simulated tape speed. If longer loops are desired, additional memory can be purchased for System V. The addition of extra memory to the system will also significantly increase the maximum tape echo time. With the variable speed option, a simulated tape loop can be transposed up or down over a range of 3 octaves.

While an actual tape loop will almost always contain an objectionable click at the point of its splice, the software of System V incorporates a special feature, Smoothsplice, which prevents such clicks from occurring. When the Freeze bar is struck, Smoothsplice searches the input signal for the optimum point at which to create a perfect, click-free splice. Striking the Freeze bar again will return the system to normal signal processing.

Once Tape Techniques begins execution, the video terminal will continue to display the data for the configuration originally selected. In order to allow the system's program to run at maximum efficiency, configuration changes made during execution are not displayed on the video terminal screen unless they are specifically requested. At any time during execution, the user can strike the "display" key on the terminal keyboard and see on the screen the data for the current configuration. While the program responds to the display request, it momentarily stops audio signal processing.

It will very often happen that the user will wish to experiment with a number of Tape Technique configurations, searching for one he particularly likes. System V allows the creation of a library of up to 20 different configurations, any one of which can be recalled during execution.

In order to store a particular configuration, the user strikes the "store" key on the video terminal keyboard. The system stops signal processing and responds:

NAME OF NEW FILE?

The user responds by typing in any five characters, including the space bar. When the fifth character is typed, the current configuration – placement of playback heads, feedback factor, and tape speed – will be automatically stored along with the new file name and the system will resume signal processing.

In order to recall a particular configuration, the user strikes the "recall" key on the terminal keyboard. The system again stops signal processing and lists on the screen all named files currently in its library, for exmaple:

1. Mary
2. 12345
3. !!
4. Aug. 9

When the user types in the number of the file he wants, its data will be displayed on the screen and the system will resume execution.

If the library is full (that is, if it already contains 20 files), the message:

LIBRARY IS FULL

will appear when the "store" key is struck. The user can then create space for a new file by striking the "erase" key. The system will list its library of files, and the user then types the number of the file he wishes to remove.

If Soundspace is selected when System V is turned on, the following list of options appears on the video terminal screen:

SELECT PROGRAM
1. AMBIENCE
2. AMBIENCE II
3. DIFFUSE REVERBERATION
4. LONG REVERBERATION
5. TIMBRE REVERBERATION
6. METALLIC EXPANDER
7. BRASS
8. ELECTRONIC STRING ENHANCER
9. WORDS
10. PERCUSSION

The user responds by typing the number of the program he wishes to use, and System V confirms his choice by erasing the screen and displaying the name of the selected program. Execution of the selected Soundspace program begins automatically.

Each program in the Soundspace collection is designed to intensify and enrich sound in a unique way. Soundspace controls the spatial feel and timbre of sound to achieve dramatic transformations of sound that were unavailable before. As more programs are developed, they will be made available to System V users. Here are descriptions of the programs currently included in Soundspace:

1. Ambience: A combination of digital delay and subtle reverberation to open up the sound of a close microphone technique.

2. Ambience II: A more pronounced version of Ambience.

3. Diffuse Reverberation: Smooth decay of 1 second.

4. Long Reverberation: Reverberation with a decay time of 15 seconds. Sounds are layered on top of sounds creating beautiful blends that slowly fade away.

5. Timbre Reverberation: An extremely bright room is simulated. The dimensions of the room can be dynamically controlled from the video terminal keyboard. Each key is assigned to a different room size; each key creates a unique timbre allowing a sound to be harmonically changed by the user. For example, a flat percussive sound can be transformed into a live pitched instrument.

6. Metallic Expander: Acoustics are simulated that emphasize the metallic qualities of instrumental sounds.

7. Brass: When patched into a synthesizer, creates big reverberant brass sounds from simple enveloped sine waves.

8. Electronic String Enhancer: Makes string synthesizers sound more natural.

9. Words: Acoustics designed for vocal tracks.

10. Percussion: Emphasizes resonant percussive decay. Each key on the video terminal keyboard creates a unique percussive resonance.

Normally, it will not be necessary to reload System V's memory, since magnetic core is used to store the software, and core retains programs even when power is completely turned off. However, it may occassionally be necessary to reload software because of severe electrical disturbance, mechanical shock, or momentary hardware failure. Also, as Computer Music develops new software for its customers a procedure for loading it will be necessary. Therefore, System V contains a program in read only memory that allows the system to be reprogrammed from specially encoded audio tapes. These tapes can be played into System V from a recording studio's standard tape machines.

We have, then, a digital audio system that can be used to create a wide range of effects with only changes in its programming. It is the first system to be made commercially available to the professional audio industry that permits easy utilization by non-technical personnel and operates entirely in real time. This description is presented for the information of the reader and as an indication of the direction in which advanced digital techniques will probably be taking audio technology in the future. It may not be too long before duplication of program processing capability many times over, as in the case of present-day multi-track consoles, will be replaced with more sophisticated software and additional memory rather than with more buttons, knobs, channels and meters.

Although it is hazardous to attempt predictions of the course of the technological future, it would be relatively safe to suggest that change will be a more prominent factor in professional audio than it has been in the past. Systems that can be changed by programming as opposed to hardware replacement cannot help but make the world a lot different from the way it used to be.

Among the credits on an increasingly larger number of albums these days is this prominent indicia:

APHEX PROCESSED TM

Aphex . . . after some two years of use is still largely a mystery to many segments of the recording and sound reinforcement industries. The following article by Howard Cummings substantially analyzes the seemingly ethereal Aphex . . . the psychoacoustic phenomena it exploits . . . how it works . . . how it affects audio signals . . . how it has been used . . . and, a few user comments . . . editor–

by Howard Cummings

After many months of conjecture and speculation, Aphex Systems has finally released information on the Aphex Aural Exciter, a unit that has been used on a number of live concerts, TV shows, movies, and recording projects for almost two years.

OVERVIEW: WHAT IT IS

The device is called an "exciter" because it does indeed tend to excite certain aural functions when used, lending additional *presence* to both recorded and sound reinforced material. The resulting augmentation interpreted by the hearing system leads to swift apparent alterations in directional information, which in turn spreads the horizontal audio picture and thus creates a new spatial illusion.

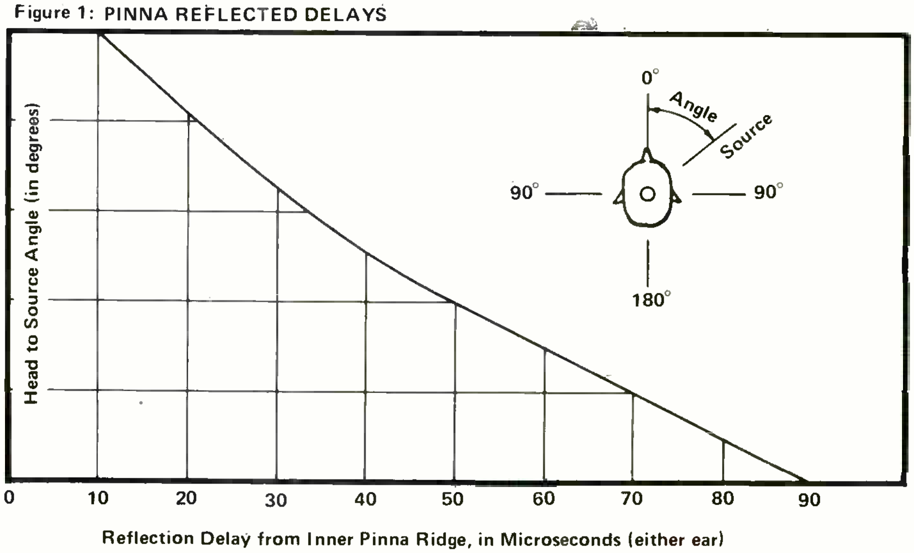
It should be clarified at the start that original audio stimuli are *not* created by Aphex, but are only further "heightened" or "stimulated" by the device. In short, this stimulation is achieved through an exclusive encoding process which consists of a low-level (-15 to -30 dB) sub-carrier signal that is mixed with original source material. This can be accomplished in two ways: 1) Acoustically, with a speaker and amplifier system for Aphex output using a separate speaker-amp system for un-Aphexed output, or 2) Electrically, where the Aphex output is combined *before* the amplifiers and with source material similar to the way a reverb mix is accomplished. The only decoding that is necessary takes place through normal hearing. *There is no auxiliary decoding equipment required.* More on the workings later.

PSYCHOACOUSTIC REVIEW

The study of psychoacoustic phenomenon is nothing new. A great deal of work has been done to perfect linearity, distortion specifications, and the development of transducers – everything to insure that any sound that is recorded is reproduced faithfully. But until recently, little attention has been paid to what happens in the receiving end of audio – namely the listener himself. When a listener hears any sound, be it acoustic, reinforced, or recorded, he is always subject to psychoacoustic interpretations of sound pressure variations, atmospheric conditions, and his own hearing limitations and processing capabilities – the ability of the brain to faithfully inform his consciousness of the stimuli received. Whatever processing takes place, up to and including the speakers, is really only half of a system. The other half neither Aphex or any other but the Supreme equipment manufacturer designed – although they all take advantage of it – namely, the human auditory apparatus.

The sequence of events that takes place between the outer ear and our perception of what is being received contains several processes – some of which are quite mechanical (as in the ear canal), and some of which are mysteriously linked to the functions of the brain, and of which more would like to be known. One of these is the sum and differencing principle, and since the sum and differencing principle is inherent in human hearing, it is used to advantage in the Aphex concept. The brain works constantly with this differen-

Figure 1: PINNA REFLECTED DELAYS



tial-comparison information (sum and difference, left and right, top and bottom, front and back, direct and reflected, ambience). Essentially, then, Aphex artificially supplies an additional differential chain of information relative to the main signal. This low-level chain of information (-15 to -30 dB) is sent out, differentially related to the main source, and is so interpreted by the brain that it ends up as the "Aphex effect".

To continue, even slight phase or frequency differences existing in binaural information being channeled into the hearing center will make the brain convey not only the primary frequencies, but also the sum and difference of the two frequencies. (e.g. If we take 1,000 Hz. and 1,100 Hz. and channel them discretely to the left and right ears respectively, we will perceive 1,000 Hz., 1,100 Hz., 2,100 Hz., and 100 Hz. The 2,100 Hz. will be perceived at a lower level, and the 100 Hz. will be perceived at a stronger level, possibly because of the inherent strength of lower frequencies and because the differencing frequencies are always more prominent in this psychoacoustic principle.)

A bit more psychoacoustic review: Our two ears act discretely and the output of each is transmitted to the "mixing center", where it is processed. If two frequencies, A & B, are isolated and "A" channeled into the right ear and "B" into the left ear, they are combined internally and the hearing center will hear and combine both in the "mixing center" of the brain to give us a composite picture – even though they were not combined externally before reaching the head.

Another concept involved is the short-term delay system of hearing, the "erasable loop" (short-term memory), that helps to relate a frequency to itself in order to discriminate against itself. (e.g. If a person were to perceive a "1 kHz. tone", he would not be able to call it a 1 kHz. tone without having a sufficient sample to compare it to. Such a tone would be meaningless without that comparison.)

Pinna (ear flaps) delays (Figure 1) also influence the hearing process. The brain interprets inner pinna delays under 100 μsec. as horizontal information and outer ridge delays over 100 μsec. as vertical information. To illustrate an example in processing: If a person were looking out a window and hears a paper rustling to one side of him, his brain receives two delay patterns from the two pinna ridges. Related to the zero-axis of the head, the brain will interpret the movement and process the azimuth of both angles to plot an intersection of the two points. It would then tell him from the two phase-delay bands, that the sound is emanating from 30° in the horizontal plane and 30° in the vertical plane.

Another factor which must not be overlooked, and the last to be discussed here, is the area of "mental coloration" – the ability or inability of the brain to faithfully inform the hearing center of stimuli received. In this context, coloration is defined as any alteration or deviation from the original source. In the brain there is considerable manipulation-distortion that routinely occurs in the hearing center, one of them being that the brain establishes hearing priorities by screening various audio stimuli and having the body react accordingly. This "prejudice-screening" is developed over the course of a lifetime and reacts to irritating stimuli such as scraping on a blackboard or glass.

APHEX – What It Does, How It Does It:

It is important to state that Aphex works primarily in the mid and high frequency ranges to increase intelligibility. To avoid high-energy, low-frequency processing, a high-pass filter of 500 Hz. (12 dB/octave) is used. It is equally important to emphasize again that Aphex is not meant to be listened to alone. If an Aphex

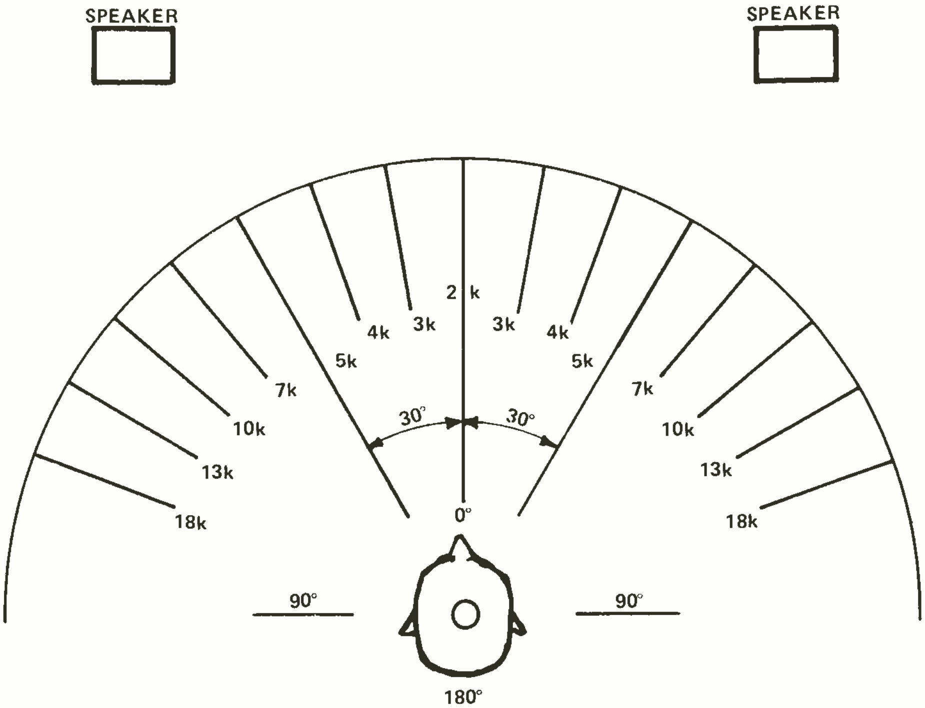
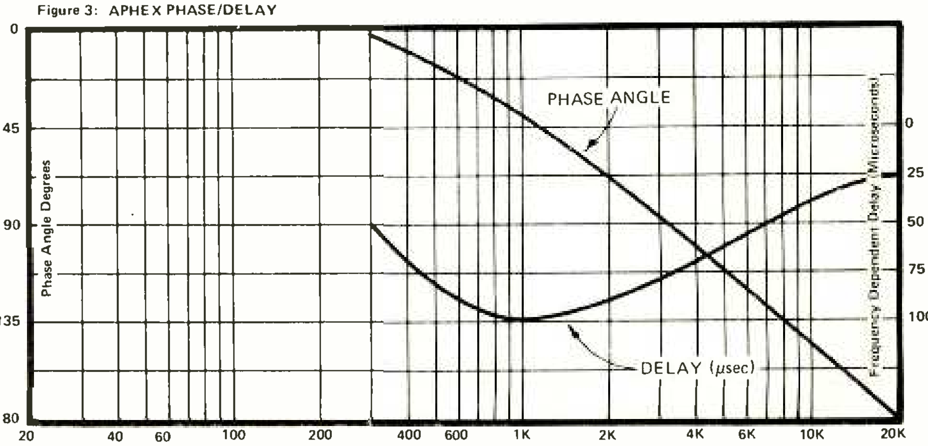


Figure 2: STEREO EFFECT – Superimposed Aphex Delay
Patterns spread both left and right images by frequency contents, as shown. Real-time image placement by Pinnae delays is 30° off axis for each ear placing speakers in their proper positions by causing approximately 60μsec delay patterns

Figure 3: APHEX PHASE/DELAY



E.G. – 1 kHz is approximately 36° out of phase and has a 100μsec. delay
– 7 kHz is approximately 130° out of phase and has 50μsec delay


Broken strings
Lost plectrums
Ripped skins
Limp stands
Pipless maraccas
Crackling leads
At least you don't have to worry about the mixer.
Soundcraft Series 1.
World's first road mixer built into a strong metal flight case.
Turn it off, shut the lid, throw it in the back of the truck.
It'll work perfectly tomorrow. And tomorrow and tomorrow, for years to come.
12 into 2 or 16 into 2. Three mixes; main stereo, monitor and echo.
Each input channel has variable line/mic attentuation, 4-band equalisation, peak level LED indicator, echo and monitor (foldback), pan, mute and pfl.
Each output has 2-band equalisation, (so does the monitor master) and the illuminated VU meters show master, monitor or pfl.
THD<0.1% Max mic gain 70dB. Relative input noise -125dBm.
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SOUNDCRAFT ELECTRONICS LIMITED

signal is soloed, one would find that the sound is unimpressive — it being very low level and lacking in bass content because of the 500 Hz. roll-off. It *is* meant to be a companion to, and combined with an un-Aphexed signal source. The un-Aphexed signal should thus contain all of the bass content that would be desired. The fundamentals of the bass are affected to only a slight degree, but the harmonics are affected more so. (Namely those in the 2-3 kHz. range where the plucking and attack of bass strings, for example, takes place.)

The Aphex signal that is mixed with source material is referred to as the sub-carrier, and is not in phase with the main carrier. This is intentional. The out-of-phase existence varies with different frequencies (Figure 3) so that a 1 kHz. tone may yield a phase lag of 36°, 5 kHz. would yield 110°, etc.

The desireable phase-shift that takes place occurs in a controlled random manner in that different frequencies are processed with precise amounts of phase shift. In the case of the wide interaction of different frequencies in music, a randomizing of phase occurs, albeit in a precise manner. Again, 1 kHz. reacts in one way, 5 kHz. in another way. (Figure 4) This leads the psychoacoustic circuitry of the body to react to the difference between Aphex and un-Aphex program material, and respond to the differences in enhancement of presence, intelligibility, and quality.

The random phase characteristics of Aphex related to the source turns out to be most useful. In addition to mimicing delay patterns otherwise created by ear pinna (random phase), it also makes the sub-carriers practical. If main and sub-carriers were in a perfect in-phase relationship, the much weaker Aphex imprints would become absorbed and would lose their independence and unique identity, and for Aphex to add to the dimension by *apparently* widening the source, it is creating an ambience effect that encompasses the listener instead of emanating directly from the source.

Aphex essentially affects audio in two areas, both equally important. Consider that acoustic instruments tend to project

APHEX: DELAY IN MICROSECONDS AND MILLIMETERS RESULTING FROM PHASE ANGLES

Hz.	*Cycle Duration Microseconds*	*Phase Angle Degrees*	*DELAY (Microseconds)*	*DELAY (Millimeters)*
300	3,333	5	46	15.9
400	2,500	10	69	23.8
500	2,000	15	83	28.6
600	1,666	19	84	28.9
700	1,928	26	91	31.4
800	1,250	29	94	32.2
900	1,111	31	96	32.5
1,000	1,000	36	100	34.5
2,000	500	67	93	31.9
3,000	333	85	79	27.0
4,000	250	100	69	23.8
5,000	200	110	66	21.0
6,000	166	119	55	18.8
7,000	143	126	50	17.2
8,000	125	134	46	16.0
9,000	111	140	43	14.8
10,000	100	145	40	13.8
15,000	66	160	29	10.1
20,000	50	170	24	8.1

a great deal of important but fragile ambient information which is always present, but tends to become "washed-out" as it is processed electronically (amplified, EQ'd, etc.) — the fundamental of which then becomes more prominent as the ambient of the instrument and the room deteriorates quite rapidly. Thus it is that the first affect Aphex can have is that of enhancing and restoring some semblence of the ambience to its natural state. Ambience is actually a three dimensional concept . . . and for Aphex to add to the dimension by *apparently* widening the source, it is creating an ambience effect that encompasses the listener instead of emanating directly from the source.

The second area that Aphex influences is the purposeful generation of harmonics from acoustical sound components in order to amplify and enhance the second and third order harmonics (human voice, piano, percussion) and their interaction, thereby adding a new dimension. These second-order harmonics are especially important since most solid-state amplification is lacking in this quality.

It is important to remember this effect only appears as an imprint on the sub-carrier -15 to -30 dB below program content, and does not affect the integrity of the main signal.

USAGE

As we have seen, a symbiotic relationship exists between the main signal and the Aphex signal. One relies on the other to enhance it and the Aphex signal is a meaningless entity without being compared to the main signal. The main signal can exist without Aphex, but Aphex cannot exist without the main signal.

Mix-Assign Procedures

Two Aphex channels are used per stereo mix. By using the unit in the foldback circuitry (echo sends and receives, cues, etc.) of the console, a selective amount of a specific channel can be assigned to the input of the unit through the "send" facility on the board. The Aphex will then "see' what has been assigned to it.

By acting as a temporary storage unit, Aphex receives information and then sends it back to the recording console. The amount of information sent to it is up to the mixer. The amount of Aphexed material *re-assigned* to the track is again up to the mixer, establishing two determinations: 1) *What* he wants to Aphex, and 2) *How much* Aphex he wants to use, both individually and overall.

Effect on EQ

It has been found that the need for a great deal of mid and high frequency EQ is lessened by incorporating Aphex into a finished track. But this should not preclude the mixer from adding EQ where needed, since Aphex is not the cure-all of recording any more than any other piece of hardware. Then which comes first — adding Aphex before EQ or vice-versa? In the case of working with deficient tracks, experience has shown that these should be processed in the way one normally would. EQ first then add the amount of Aphex enhancement desired, within reason.

As a caution it should not be used on a sibilant vocal because of the undesirable harmonic enhancement that would take place within the Aphex bandwidth. Because of potential sibilance, the unit has a built-in notch filter at 5kHz. (-14dB) to counteract this.

How Much Aphex

By adding Aphex, enhancement takes place without adding a significant amount of level. It is said, if you can measure a level difference by adding Aphex, you have used too much. There is an increase in transient peaks, but not in average level, and, the difference exists only in *apparent* level because of the psychoacoustic reaction taking place.

The effect of this apparent level increase is dramatic. As an example, it has been noted that in a sound reinforcement situation, one could experience approximately a 6 dB apparent level increase. This could be measured only indirectly, since obviously you cannot place a VU or SPL meter inside a person's head. But it can be calculated by removing Aphex from the system and then measuring and adding the amount of level that existed before Aphex was removed. A test of this sort has been conducted by Aphex Systems, and it was found that a difference of 3-6 dB apparent level existed in a pre and post situation. If this is the case, think of the amplification power that could be saved in the mid and high frequencies in sound reinforcement applications!

In sound reinforcement, using Aphex on deficient signals (such as acoustic piano with a Countryman or Helpinstill pickup) can enhance tonality, definition, and restore missing ambience to the instrument for both on-stage performers and the audience.

User Comments

Typical of comments heard concerning Aphex in a sound reinforcement application were those from Ed Kolakowski of Sound Productions, who used the unit during the 1977 Led Zeppelin tour during which he employed it for keyboard stage-monitoring for Zeppelin member John Paul Jones. These comments are followed by those of Lew Mark, independent recording engineer using Aphex with the contemporary rock-jazz group, Kalapana.

Ed Kolakowski — "This tour isn't really keyboard oriented except for NO QUARTER. On that song he uses a Fender Rhodes and the Steinway and we use Aphex in his stage monitor for clarity more than anything else . . . and definition I guess you could say — it really helps that.

I know Zeppelin also uses it in the vocal monitors and it's beamed down to the rest of the members from the "flying monitors".

Q: When did you decide to use it?
A: It was Showco's decision.

Q: What were your experiences?
A: Paul McCartney & Wings on vocals and Grand piano. We achieved a piano sound I don't think anyone has gotten in a long time. The unit was also used on the James Taylor tour in the summer of 1976.

Q: How do you use it?
A: I assign the main source and the Aphex unit to the faders and mix each independently according to Jones' (visual) signals.

Q: What do you think it does? How is it beneficial?
A: It clarifies a lot of things. You don't have to run things as hot as you would without it. It has a very good carrying power — it really lets the track cut through. It has very good definition, articulation, and depth. Things don't get "lost" with each other — there's a good blend, yet good depth and separation between instruments.

User Comments In Studio Application

Lew Mark — Independent recording engineer, Los Angeles, using the unit with the group Kalapana, a contemporary rock-jazz group.

Q: When did you decide to use it?
A: I was introduced to the unit at the Santa Monica Civic Audjtorium (L.A. entertainment venue) by one of the Bellamy Brothers. I tried it out in the P.A. system but couldn't get an idea of what the thing was doing because of the lay-out of the Civic. So I decided to try it out on my next album project.

Q: What were your experiences?
A: All good. It's hard to put it into musical terms. Aurally it created a sensation that enhanced clarity and . . . presence I guess the word would be. When I had finished this album with the unit, I had become surprised with what I had been listening to in the past. They all seemed muddy compared to what the Aphex signal did.

Q: How do you use it?
A: I used it in the mix of the album. I tried it with all instruments, but my main concern with this group was vocals. If there was any device that claimed to make vocals clearer, more precise, more . . . *articulate*, then I was going to try the device regardless of the cost. I was also told by the manufacturer to try it on other acoustic instruments because it would have a great enhancing effect on harmonics. I tried it and it sure did. I used it on strings, electronic and acoustic pianos, and just about every instrument that could handle it. The saxophone was a bit unpleasant with Aphex — it made the harmonics sound a bit too harsh — and most of the drums couldn't take but a fraction, but I used it on the snare for a tiny bit of brightness.

When I mixed this LP I assigned the Aphex signal directly to center even though it was a stereo project. Everything brightened up and it seemed to place instruments more where they belonged. I returned it to center (mono) because channels A and B were identical. There was a three-part vocal thing done into three mikes with three placement settings. I placed the vocals left-center-right and brought back the Aphex signal to center. The Aphex signal didn't seem to have any *noticeable* effect on the stereo placement. I presume that the pan assignments were being affected but because of the psychoacoustic phenomena Aphex is providing, it causes the vocals to come out of other places, as certain frequencies are reached, that I would never be able to get out of equalization. Therefore the vocals came out of where I assigned them, *but* with some kind of brilliance emanating from all over dependent on frequency.

So whereas most mixers use the unit in stereo, I used it monaurally, because I wanted to pad the center image and it worked out very well this way. It didn't seem to have any affect on the sides, and the perception of left and right didn't seem to change, it even seemed to get stronger perhaps because the center image was more defined than it was previously. In future albums, I'll experiment with other techniques and I don't think I'll ever settle on one.

Q: What do you think it does?

A: There's more of a presence instead of an image-shift — it's more of a *psychoacoustic* image-shift but not a *physical* image-shift. I didn't notice in mix-down that the image shifted any less by bringing up the Aphex mono signal to center. The only answer I have is that because the center is strengthened, relating to where the center lies for the listener, the left and right become more strongly aligned.

One interesting technique I tried was to phase-shift the Aphex with an Eventide Flanger which gave me a very unusual sound on one of the cuts. So I stereo-flanged a mono Aphex signal. I'd never be able to describe the final effect. When you take a regular note and flange for out-of-phase combing effect and then take the Aphex signal, which is something like 180° reversal, different frequencies are out of phase at different degrees. It was a matter of using a phase-differential producing device and phasing it.

Q: Is it beneficial; how?

A: First of all, it frees the EQ to do other things. Most boards are graphic EQ. When you have a chance, you can use parametric on some of the channels, But you have to use one knob to perform two or three functions. You might be able to add at 10 kHz., but that knob may also be the 5 kHz. knob. So you have to make a choice.

The second thing is it adds an extra "dimension". If you play with just the Aphex signal itself, increase of that signal will cause a very strange perception. The sound will appear to emit from different places depending on frequency — that is, in a stereo situation, if you close your eyes, the more you increase the Aphex signal, higher frequencies seem to emanate from outside the speakers from different points, primarily because of the psychoacoustic phenomena.

Back to EQ. I didn't have to over-EQ things to get the same sounds that I used to be getting. By backing off, almost to nothing on my EQ, and using Aphex instead, I received more of a "broad-banded equalization". I tried, on one vocal passage, to use just equalization and not Aphex and I wound up using 12 dB at 10 k to get a *similar* effect. I A-B'ed, I spent hours. It wasn't the same — it didn't work. So it's beneficial with the EQ factor plus adding that extra "dimension".

It's not going to make a bad mixer good. Nothing can replace good work, talent, and perseverance on your mix — this is just another tool. □□□

New Products

GEMINI, NEW AUDIO & DESIGN RECORDING STEREO COMPRESSOR/LIMITER

Having most of the flexibility of more exotic units, while retaining a simplicity of operation, the compact Gemini is said to be particularly ideal for the requirements of the self-op recording artist as well as other large and smaller recording studio applications.

The Gemini may be used for stereo or dual mono compression/limiting. The system is suitable for operation with all systems having levels between -10 and +10 dBm. The output limit threshold can be simply pre-set to suit following equipment. It is then only necessary to select limit or comressor function (in the compress mode you can select either 1.5:1 or 3:1) and just increase the input potentiometer to obtain the desired amount of compression or limiting. An added feature is that after 10 dB compression the slope tightens to that of a limiter so that unexpected peaks cannot be sent into the following system.

A light indication system to show compression and limiting was selected by the designers for the ease with which these can be seen from a distance, contrasted to the more conventional meters. The orange indicator just lights as the threshold of compression is approached; as it comes to full brightness the red indicator lights at 10 dB gain reduction, to go to full brightness for 20 dB.

An in/out switch allows each channel to be bypassed allowing direct comparison between direct and the processed signal.

On voice the compressor/limiter combination makes use of a soft slope to retain dynamics while providing the ability to hold sudden peaks as they occur. Bass instruments can be tightly controlled, keeping difficult peaks down to reasonable level, yet increasing the overall impact of the sound through higher average levels. Mid-band instruments can be compressed before or after equalization for maximum intensity and bite. In sound reinforcement applications power amplifiers can be dramatically increased in efficiency while still providing speaker protection when fed from the Gemini, while direct injection from an electronic instrument amplifier will give considerable scope for on-stage use, without fear of feedback.

Audio Design Recording, Inc., is the U.S. distribution company for the products of Audio & Design Recording, Ltd., in Great Britain.

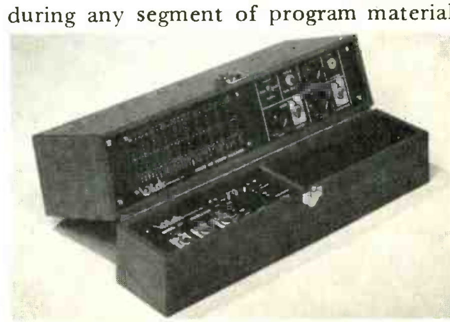
AUDIO & DESIGN RECORDING, INC.
P.O. BOX 23047
HONOLULU, HAWAII 96822

for additional information circle number 47

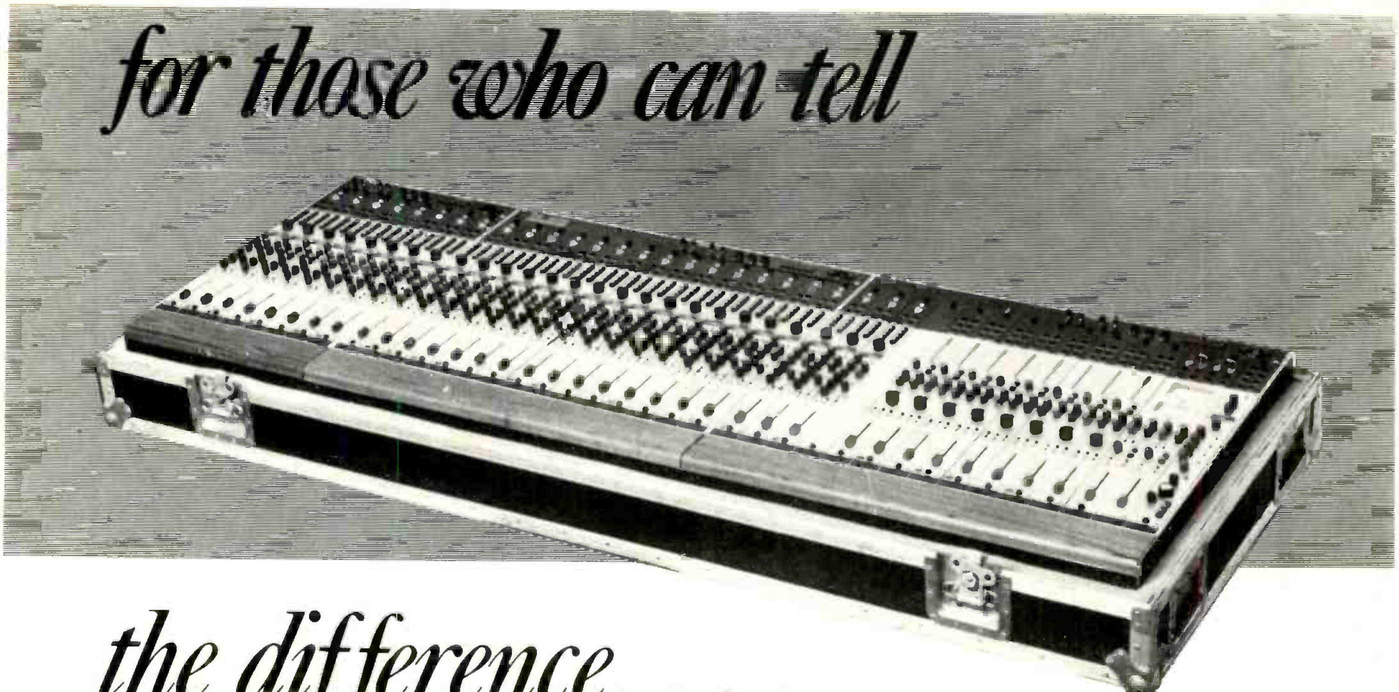
WHITE MODEL 142A SPECTRUM MONITOR

The model 142 A Spectrum Monitor is a peak reading real-time analyzer designed to be used primarily as a monitor for program material. The unit incorporates a 28 by 11 LED matrix for the display of 27 one-third octave channels from 40 Hz to 16 kHz plus one broadband channel for overall indication of level. A front panel switch selects display ranges of 10 dB or 30 dB. In addition, the input is calibrated in 10 dB steps from -40 dBm to +10 dBm or 50 dBspl to 100 dBspl. Each one-third octave channel is peak reading with a decay time of 1/2 second.

Two CMOS memory registers allow the engineer to store display information. This information remains stored as long as the unit is turned on. In the SAMPLE mode, "snap shots" of program material can be stored in either memory for later viewing and comparisons. In the ACCUMULATE mode, either memory can be used to register the highest peak readings during any segment of program material



for additional information circle number 48



the difference. . . .

The TRIDENT FLEXIMIX portable mixing system provides the comprehensive facilities which would normally be found only on an expensive studio console. This gives Fleximix the big desk 'feel' as soon as you operate it. Add this to the unique expandability of the system and you can see why it is in a class of its own. For a little over $4,000 you can buy a 10 input–2 output configuration, which could subsequently be expanded to a system with 10 mixed outputs, any number of input channels and 24 track monitoring. Expansion is simply achieved by slotting-in additional channel modules. When available slots are used up, another mainframe is added. Modules may be placed in any sequence. No factory rework or rewiring is necessary. Additional mainframes may be either rigidly or flexibly coupled to the original system and flight-cases are available to accommodate any arrangement.

A number of exciting new modules are now available which will extend even farther the system versatility. These include a Compressor/Limiter module. Quad joystick module and Line-Balancing module.

Send for details to:
Trident Audio Developments, Ltd.
Sales Office: **112/114 Wardour Street**
London W1V 3AW
Tel: 01-734-9901. Telex 27782 Tridisc.

Factory Address:
Shepperton Studios
Squiresbridge Road
Shepperton, Middlesex.
Tel: Chertsey (09328) 60241.

TRIAD

United States Agent:
Studio Maintenance Service
12438 Magnolia Boulevard
North Hollywood
Ca. 91607
Tel: (213) 877-3311
Contact: David Michaels

Canadian Agent:
Audio Analysts, Inc.,
2401-A St. Catherine St. East,
Montreal H2K 2J7, Quebec.
Tel: (514) 525-2666. Contact: Pierre Pare

for additional information circle number 49

and viewed later to ascertain whether or not some maximum level was surpassed.

Suggested applications for the Model 142A Spectrum Monitor include program material monitoring, recording and mixdown analysis, monitor system adjustment, tape equalization and calibration, frequency response testing, and many others. Of particular value will be the dual memories for before/after camparisons.

The Model 142A is 3½ inches high by 8 inches deep on a standard rack panel. The unit weighs approximately 8 pounds. The front panel is black anodized brushed aluminum. Either 115 or 230 VAC power at 15 Watts is required.

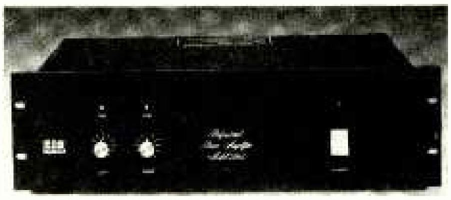
The suggested list price for the Model 142A ia $2,500.

WHITE INSTRUMENTS, INC.
P.O. BOX 698
AUSTIN, TEXAS 78767

for additional information circle number 50

NEW MODEL 250C AMP FROM B G W

BGW Systems announces the introduction of it's new professional power amplifier, the Model 250C. Delivering 100 Watts/channel into 8 Ohms with less than .03% IM distortion, and 150 Watts/channel into 4 Ohms at 1 kHz, the 250C features the following:



Modularized amplifier output stages; front panel clipping indicators; relay operated delay and speaker protection; front panel gain controls; front panel mounted magnetic circuit breaker; separate chassis and signal grounding; rear panel connections for elimination of ground loops; rear panel convertible bridged mono operation slide switch (mono output power: 251 Watts into 8 Ohms); ¼"TRS and Cannon type input connectors; redundant, fail-safe, heavy duty transistors; and plug-in matching input transformer provision with mechanical guard.

Brief specifications include: Hum and noise level, better than 110 dB below rated output into 8 Ohms. The unit measures 5.25 inches by 19 inches standard rack front panel, by 11.75 inches deep.

The professional net price is $559.00

B G W SYSTEMS
13130 YUKON AVENUE
HAWTHORNE, CA. 90250
(213) 973 8090

for additional information circle number 51

ALLISON RESEARCH SERIES 65K AUTOMATION PROGRAMMERS

The 65K series programmers are second generation devices which provide the memory function for programmable mixdown consoles. Through the use of priority encoding techniques, essentially unlimited capacities of up to 65,536 bits, 8,192 analog functions are possible.

A negligible access time of 3.2 milliseconds is imposed, regardless of the number of programmed funtions, allowing extremely complex consoles to be automated, while maintaining an instantaneous response to operator movements. Four levels of error detection assure absolute validity of decoded data, regardless of the condition of the storage medium.

The 65 K is available as a direct replacement for first generation automation equipment, or as orignal equipment for console manufacturers. It is also available in conjunction with the Allison Research Memory Plus system, as an outboard, portable automation system which may be patched into any existing nonautomated console.

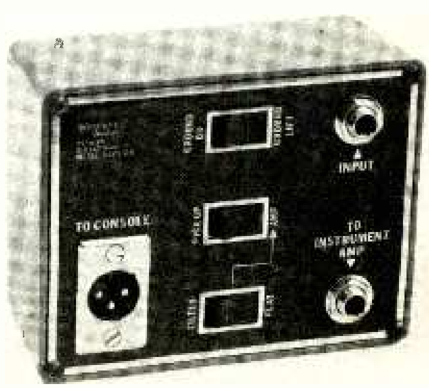
Price of the 32 function analog programmer No. 65K-A1-32 is $4,246.00. Delivery is within one to two weeks.

ALLISON RESERCH, INC.
2817 ERICA PLACE
P.O. BOX 40288
NASHVILLE, TN 37204
PHONE: (615) ALLISON

for additional information circle number 53

WESTLAKE MODEL DBP-1100 DIRECT BOX (PASSIVE)

The DBP-1100 allows the recording engineer to record directly from the amplifier or pick-up of an electronic instrument. Input and instrument jacks are muted to facilitate simultaneous recording and instrument amplifier operation.



for additional information circle number 52

'One of the most important breakthroughs in the history of recording.'

First installation of MCI's new JH-50 Series automation system in the United States is at Criteria Recording Studios. Looking on as owner Mack Emerman explains a feature are (L to R) Dennis Bryon (Bee Gees); Karl Richardson (Bee Gees co-producer); Maurice Gibb and Blue Weaver (Bee Gees); Mack; Tom Dowd (Atlantic Records, producer of Lynyrd Skynyrd); Ronnie Van Zandt (Lynyrd Skynyrd); Barry Gibb (Bee Gees) and Albhy Galuten (Bee Gees co-producer).

Helping Hands: the automation system that works.

It may be some time before all the advantages of MCI's computerized mixing are fully realized. But Ron and Howard Albert of Fat Albert Productions are discovering new things about the system every day. "Its capabilities are almost unlimited," says Ron. "You can do **anything**. Remix as many times as you want, make each one different, and keep all of them. Lay in a new track a month later—without a click or pop or a speck of difference."

"You have total recall of every fader setting from start to finish," adds Howard. "And a 'Plasma Display' visually shows you the changes you've made without the faders moving. We think the system is one of the most important breakthroughs in the history of recording."

Ron & Howard

The easy-to-use, low-cost Helping Hands automation system is installed easily on all MCI consoles, and is also available for use with other consoles through minor modifications. Ask your local MCI dealer today about adding this remarkable capability to your present equipment.

MCI

4007 N.E. 6th Avenue, Fort Lauderdale, Florida 33308 • (305) 566-2853 • Telex 51-4362

for additional information circle number 54

The box can sit on its rubber feet on any flat surface or snap onto a mike or music stand using the optional clamps that are provided.

In keeping with Westlake Audio's second generation of studio products, total printed circuit construction and high quality components make this unit sturdy and maintenance free.

Unit specifications include: Input impedances, 20 Hz to 15 kHz greater than 50 K Ohms, 1 kHz to 5 kHz greater than 100 K Ohms.

Switch functions include:

Ground On/Ground Lift: connects the chassis grounds of the console to the guitar or guitar amp.

Pick Up/Amp: Inserts a pad between the input/instrument amp jacks (which are wired in parallel) and the isolation transformer. For applications where the signal source is the instrument itself.

Filter/Flat: Functional only when the pick up/amp switch is in the "amp" position. Provides high frequency compensation to simulate the response of a typical guitar amplifier speaker.

The DBP-1100 measures 3.75 x 2.25 x 4.75 inches (9.5 x 5.7 x 12 centimeters) and weighs less than one pound (2.2 kilograms).

WESTLAKE AUDIO
6311 WILSHIRE BOULEVARD
LOS ANGELES, CA 90048
PHONE: (213) 655-0303

for additional information circle number 55

INDUSTRIAL RESEARCH DIGITAL TIME DELAYS

CCD technology applied to digital time delay has enabled the development of the Industrial Research new DC-4011 Audio Program Delay and DD-4012 Sound Delay Module. The units offer 96 milliseconds of delay selectable in 4 millisecond increments and 2 independent outputs at a selling price which means exceptional value.

Controls on the DC-4011 are located on the front panel while on the DD-4012 they are recessed behind a tamper proof front panel. Dynamic range is greater than 90 dB, pre-emphasis is equivalent to 50 microseconds, and distortion is less than 0.25% at 1,000 Hz. The models are transformer isolated on both input and output. A red LED indicates peak clipping level while a green LED lights at 14 dB below clipping.

The units operate on power of 115/230 volts, 60/50 Hz. with a dissipation of less than 10 watts. Rack mount dimensions are 1-3/4" high by 19" wide by 8-1/2" deep. Nominal weight is seven pounds. Barrier strip connectors are standard. XLR-3 type connectors are optional on the DC-4011.

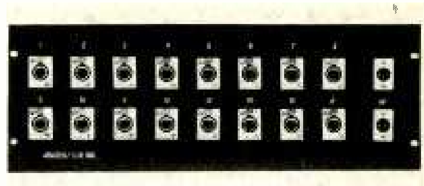
INDUSTRIAL RESEARCH PRODUCTS
321 BOND STREET
ELK GROVE VILLAGE, IL 60007
PHONE: (312) 439-3600

for additional information circle number 56

ABADON/SUN MICROPHONE INPUT PANELS

Abadon/Sun, Inc., has introduced a complete line of microphone input panels for recording studio applications. High grade aluminum stock, that has been brushed and anodized, is used for construction. The numerals are then engraved (not painted) into the panel for a beautiful and functional appearance.

The panels are available in both 16 and 24 input configurations and each comes equipped with two headphone output jacks. There is a choice of anodized black with silver numerals or natural silver with black numerals. The name of the studio can be engraved on the panel for a small extra charge.



High quality Switchcraft connectors with gold plated contacts and non-relfective matte finish are used throughout.

The 16 input version (MIP-16) sells for $129.00 and the 24 input model (MIP-24) is $155.00.

ABADON/SUN, INC.
P. O. BOX 6520
SAN ANTONIO, TX 78209
PHONE: (512) 824-8781

for additional information circle number 57

MAP NEW STEREO 100 WATT POWER AMPLIFIER

Modular Audio Products new Model 7100 Stereo 100 watt Power Amplifer is designed for professional studio monitoring and sound system applications. Measuring only 3½" high, the unit includes provisions for either 19" rack installation or bench/shelf top mount.

Featuring high impedance bridging inputs with a dual IC operational preamplifier, the design incorporates two high current, high voltage Hybrid Op-Amp modules in the direct coupled output stages. The unit provides extremely low noise, at 110 dB below rated output, and distortion of 0.1% maximum at rated output.

Stereo power output is 100 watts RMS/channel into 4 ohms, or 75 watts RMS/ channel into 8 ohms. Bridged monaural power output is 200 watts RMS into 8 ohms, or 150 RMS into 16 ohms. A switch on the rear of the chassis allows instant stereo or mono mode selection, without internal wiring changes.

Particularly useful, in high quality sound/speaker distribution and P.A. systems, is the optional internal addition of two 70.7 Volt line output transformers, requiring no additional space or external hookups by the user or system installer. Designated MAP Model 7100-1, the unit provides stereo power output of 100 watts RMS/channel at 70.7 Volts RMS into 50 ohm loads.

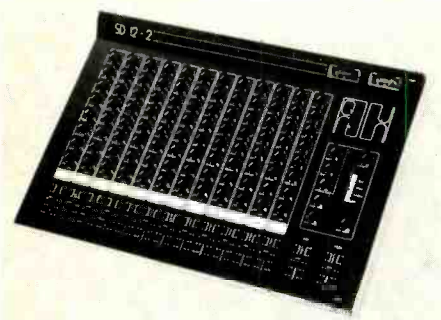
MODULAR AUDIO PRODUCTS
50 ORVILLE DRIVE
BOHEMIA, NY 11716
PHONE: (516) 567-9620

for additional information circle number 58

AUDIO MARKETING, LTD. INTRODUCES NEW A&H MIXERS

Designated the SD-12-2, this compact unit is designed for permanent or portable sound re-inforcement and recording applications.

Features include 12 low impedance microphone and line inputs; direct channel outputs; mike trim; four band equalizer; cue (or stage monitor) and echo sends; pan pot and solo on each input. The output section contains a headphone monitor circuit, and illuminated VU meters. All this comes attractively packaged in a European styled 20" x 17" x 3½" cabinet.



The low cost of this unit, under $1,000, makes it ideal for touring musicians and clubs. The output and monitoring facilities are such that the SD-12-2 can be used for stereo and four track recording, enabling musicians to produce their own demo tapes.

The S-6-2 has been designed specifically for radio, TV, film production, and disco sound booth applications. It utilizes linear faders throughout, and includes automa-

tic start and fade for fast, economical tape program production.

Features contained in the attractive 17-3/8" x 15-3/8" x 3½" console include two stereo RIAA inputs, two stereo tape inputs; two microphone inputs; equalization on all inputs; broadcast cue; stereo output plus stereo monitor output. It also utilizes a digital switching system for start-stop control of turntables or tape machines.

Economically priced at under $1,100, the versatile S-6-2 is one of the best buys on the market and in reach of even the smallest broadcast facilities. An automatic "ducking" circuit for voice-overs is standard, a feature perfect for all broadcast studios.

AUDIO MARKETING, LTD.
142 HAMILTON AVENUE
STAMFORD, CT 06902
PHONE: (203) 359-2312

for additional information circle number 59

NEW RECORD PREAMP FROM TELEX

Telex has announced a new tape Record/Playback preamplifier of unique versatility for recording studio, broadcast, and industrial applications. Designed as a modular addition to their extensive line of tape products, Telex states that the new RP-85 has wide application for single and multi-channel use in both new equipment and replacement tape systems.

The compact, monaural RP-85 will accommodate tape speeds from 1-7/8 to 15 ips in 1/8 to full track formats. A low speed RP-85L provides special equalization for use with cassette or open reel logging application (15/16—1-7/8 ips).

The unit has facilities for mixing and is universally matched for line and micro-

phone inputs as well as line and monitor outputs. Independent line and cue amplifiers permit monitoring without loading the program output. Self-synchronizing bias permits multi-channel operation including stereo monitoring. Two isolated DC outputs provide continuous or record-mode controlled voltage for external relay or lamp operation. Programmable connectors accommodate various head impedances and configurations.

The RP-85 features silent FET switching, 60 dB bias rejection and full immunity to eletromagnetic and radio frequency interference.

The new amplifier replaces the Telex RP-84, RP-110 and RP-120 units and is priced below \$260.00. Rack mount kits, interconnecting cables and 240V adapters are available as optional accessories.

TELEX
9600 ALDRICH AVENUE SOUTH
MINNEAPOLIS, MINN. 55420
PHONE: (612) 884-4051

for additional information circle number 61

AMEK CONSOLES TO BE INTRODUCED BY EVERYTHING AUDIO

Amek announces two basic versions, the Model 2016 containing 20 inputs and 16 outputs, and the Model 2816 containing 28 inputs with 16 outputs with optional 24 track metering. The chassis can be added to with individual modules at any time. The desk is built of steel and aluminum and contains a producers table with a standard patch bay containing 144 patch points. Up to 240 jack points may be housed in the area.

The console features P&G faders, 4 auxiliary sends, stereo solo in place, 4 band EQ with high and low pass filters, one mike and two line positions with adjustable gains, a pannable monitor mix with mute and solo, and panning between odd and even busses with a separate stereo mix and well as a multi-track assign buss.

The Amek also contains full control room and studio monitoring as well as self-contained test oscillator. A two track stereo master is provided along with separate stereo and mono meters.

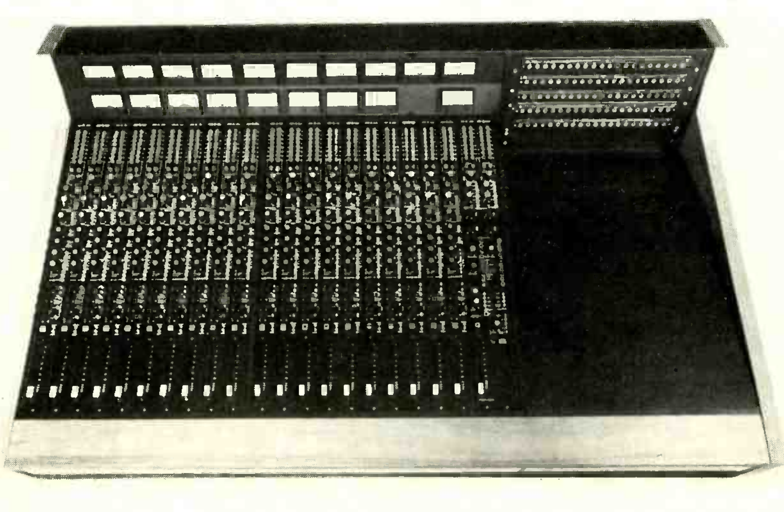
The console rests on a steel floor stand with a 19" rack bay provided under the producers table.

Maximum output specifications are +32 dB and mike input levels of +10, and line input levels of +20, are possible. The entire console is interfaced with XLR sockets and is powered by high speed 741 I.C.'s. Frequency response is 16 Hz. to 23,000 Hz.

Prices range from \$14,350. to \$20,650., F.O.B. factory. Delivery is 10 to 16 weeks A.R.O.

EVERYTHING AUDIO
7037 LAUREL CANYON BOULEVARD
NORTH HOLLYWOOD, CA 91605
PHONE: (213) 982-6200

for additional information circle number 62

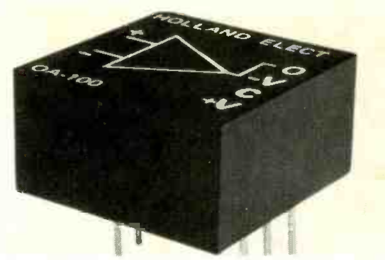


HOLLAND ELECTRONICS OA-100 OPERATIONAL AMPLIFIER

The Holland Electronics Model OA-100 operational amplifier was developed to provide an alternative to amplifier modules currently on the market.

The OA-100 features a maximum output voltage of greater than 7.75 Vrms using bipolar 16 V power. The unit is capable of supplying its full output to a 75 ohm load with less than 0.15% total harmonic distortion in a 40 dB gain configuration. The OA-100 also features less than 4 mV equivalent DC offset and a -125 dB equivalent input noise.

Large signal slew rate is specified at



greater than 8 Volts per microsecond and open loop gain at 105 dB minimum. The device is said to be pin compatible with other gain modules now on the market and is priced at \$27 each in quantities under 25 units.

HOLLAND ELECTRONICS, INC.
970 EAST 92ND STREET
BROOKLYN, NY 11236
PHONE: (212) 649-7330

for additional information circle number 63

ROCKWELL DEVELOPS REAL-TIME AUDIO PROCESSOR

A computer-based processor that eliminates unwanted noises from audio signals and recordings in real-time has been developed by Rockwell International Corporation's Electronics Research Center.

The result of several years of research, the processor represents a major breakthrough in real-time audio processing technology. "The equipment, called the Automatic Digital Audio Processor (ADAP), is a sophisticated digital computer complemented by audio processing circuits," said Cedric O'Donnell, vice president of Rockwell's Electronics Research Center.

The processor, which is being demonstrated to potential customers, can be used either to "clean" an audio signal as it is being received, or to enhance a recording that has already been made, noted Rockwell's Dr. Jim Paul. It can remove from 40 to 50 dB of highly-correlated noise with virtually no degradation in the desired voice signal.

ADAP can be used both on-site – by police in making recordings, by on-scene reporters, or for on-location television and motion picture filming – and in a studio or sound room in conjunction with other audio equipment.

The processor removes two types of noises from voice tracks: additive sounds, generally music, traffic, or other background noises; and convolutional sounds, such as resonances, room acoustics, or noises inherent in recording equipment.

"These sounds are predictable, in varying degrees, and up to 90 per cent are automatically cancelled by the processor with virtually no degradation of the voice signal," Dr. Paul explained. The equipment recognizes a foreign, or undesired, sound, and eliminates it within 200 milli-

seconds.

"We've successfully demonstrated ADAP by cleaning up a number of police voice recordings and cockpit recordings from aircraft accidents," Dr. Paul noted. "We've also used it to enhance vintage music records by eliminating the tinny sound and bad acoustics that were inherent in the old studios and recording equipment.

ADAP is easy to operate and fully automatic, adjusting itself adaptively to track and cancel undesired audio components, said Dr. Paul. No special knowledge is needed to use the processor, and it can be readily adjusted for a broad variety of audio enhancement jobs.

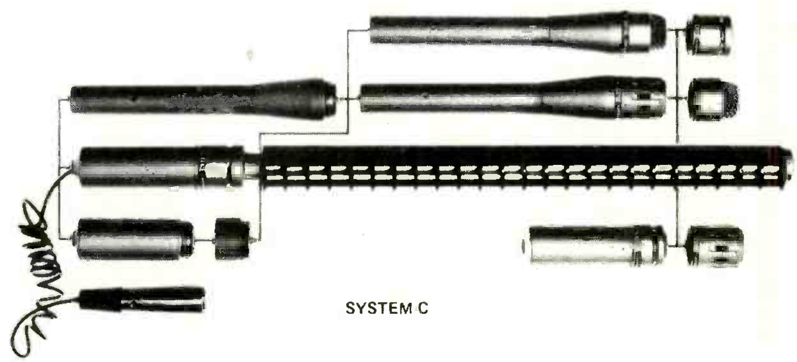
The processor has a full complement of panel controls that may be replaced by internal plug-in switches for single-use applications. In addition, it is compact and portable, and easy to utilize in conjunction with other audio equipment.

ROCKWELL INTERNATIONAL
AUTONETICS GROUP
3570 MIRALOMA AVENUE
ANAHEIM, CA 92803
PHONE: (714) 632-4195

for additional information circle number 64

ELECTRO-VOICE INTRODUCES NEW CONDENSER MICROPHONE SYSTEM

Identified as "System C", the new modular professional microphone system consists of a number of elements which can be interchanged to fit specific applications. Elements in the system include two electronic preamplifiers – one for handheld applications and one for boom use. The boom preamplifer operates from either phantom or AB remote power. Four interchangable capsules are available – omnidirectional, cardioid, hypercardioid and Cardiline® shotgun. A complete line of accessories including windscreens and shock mounts are also available.



SYSTEM C

According to Electro-Voice, "System C" microphones use a special proprietary charging process and are said to be as rugged and reliable as any dynamic microphone. "System C" is backed by Electro-Voice's two-year, unconditional warranty.

ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MICH. 49107
PHONE: (616) 695-6831

for additional information circle number 66

AUDIO-TECHNICA UNVEILS ITS FIRST MICROPHONES

Firmly established in phono cartridges and headphones, Audio-Technica U. S. has entered another area of "small transducers" – microphones.

Audio-Technica's first microphones are available in three electret condenser and two dynamic "moving coil" models. Each type is available with cardioid (unidirectional) and omnidirectional acceptance patterns.

"Microphones share technical and marketing similarities with our other product lines," notes Jon R. Kelly, A-T vice president and general manager. Worldwide, Audio-Technica has had 15 years of experience in transducer technology.

Each Audio-Technica mike has applications in broadcasting, professional and

amateur recording, live performance, public address and other activities using sound reinforcement.

A-T's initial offering of electret condenser mikes are designated AT801, AT811 and AT813. All have aluminum alloy bodies.

The AT801 is Audio-Technica's omnidirectional electret condenser microphone. Listing for $50. the AT801 yields a frequency response of 40-18,000 Hz. Sensitivity is rated at -48 dB. The output, balanced, is 600 Ohms (nominal). Weight is 5.6 oz., less cable and clamp.

The AT811 is a cardioid (unidirectional) electret condenser mike. Listing for $75, it yields a frequency response of 50–20,000 Hz. Sensitivity is rated at -54 dB. The output, balanced, is 600 Ohms (nominal). Weight is 6 oz., less cable and clamp.

At $80, the AT813 is Audio-Technica's top mike. It is an electret condenser unit with a unidirectional acceptance pattern. Its frequency response measures 20–20,000 Hz. Sensitivity is -58 dB. The output, balanced, is 600 Ohms (nominal). Weight is 6 oz., less cable and clamp.

A special feature of the AT813 is a large, permanent, built-in protective screen with multistage filters to reduce wind noise and "pops" from close-up use.

All electret models have permanently polarized diaphragms and need no external power source. One AA-size cell, preferably an alkaline or mercury cell, is placed inside the microphone to power the impedance-matching network. An alkaline cell should last approximately 3,500 hours in intermittent service.

A-T's two dynamic "moving coil" mikes, in which the coil is driven by a sound-activated diaphragm, are designated AT802 and AT812.

The AT802 is an omnidirectional unit. The body is of sturdy die-cast zinc. Specs indicate a frequency response of 50–16,000 Hz. Sensitivity is -54 dB. The output, balanced is 600 Ohms (nominal). Weight is 5 oz. The unit lists for $50.

The cardioid (unidirectional) AT812 has a frequency response of 50–18,000 Hz. Sensitivity is -60 dB. The output, balanced, is 600 Ohms (nominal). Weight is 7.6 oz. The body is aluminum alloy. The suggested list price is $75.

Accessories included with all A-T mikes are a 16-½ inch, 2 conductor, shielded cable with XLR-3 (Switchcraft A3F) microphone connector, a snap-on stand clamp, and protective carrying case.

Optional accessories include the AT-8201 Line Matching Transformer for matching mike output to systems with high input impedances (600 Ohms to Hi-Z), the AT8101, AT8111 and AT8112 slip-on foam windscreen, and the AT8410 shock mount.

AUDIO-TECHNICA U.S., INC.
33 SHIAWASSEE AVENUE
FAIRLAWN, OHIO 44313
(216) 836-0246

for additional information circle number 68

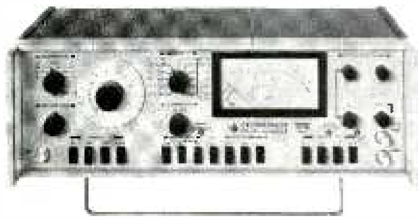
for additional information circle number 67

HAMMOND INDUSTRIES NOW EXCLUSIVE U.S. DISTRIBUTOR OF FERROGRAPH TEST EQUIPMENT

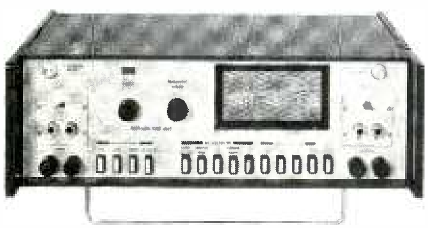
Hammond Industries is very proud to announce their appointment as the exclusive U.S. distributors for Ferrograph test equipment.

Ferrograph test equipment is the result of close collabortion between test equipment design engineers and audio engineers, and is thus specifically tailored towards the special requirements of audio testing.

For example, the RTS 2 test set, designed for general purpose audio testing, incorporates a signal generator, millivoltmeter, distortion analyzer, and wow and flutter meter, and requires only one input

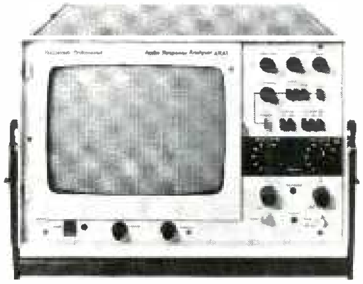


connection and one output connection to the unit under test. This facility, plus the simple and logical arrangement of the front panel controls, allows full tests of any audio unit to be made with the minimum expenditure of skilled time.



The flexibility of the RTS 2 may be further extended by the ATU 1, which provides balanced input and outputs, loudspeaker monitoring, extended level capability, and other features needed to test professional and broadcast audio equipment. The A/B monitor facility also makes the ATU 1 particularly useful for audio clinics.

Where a graphic display of frequency response is required, the Ferrograph ARA 1 provides a versatile self-contained frequency response analyzer with a large 12 inch screen readout. Precise response analysis of any audio equipment is extremely



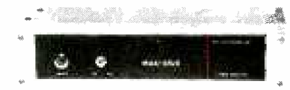
simple, and the ARA 1 may also be used with test tapes and records, with automatic synchronization. The optional four signal (simultaneous display) digital store, available shortly, provides the ARA 1 with facilities not available even on equipment at many times the cost.

HAMMOND INDUSTRIES, INC.
155 MICHAEL DRIVE
SYOSSET, NY 11791
PHONE: (516) 364-1900

for additional information circle number 69

RAMKO MICROPHONE DISTRIBUTION AMPLIFIERS

Ramko Research, Inc., announces the availability of a series of microphone distribution amplifiers that enable the user to amplify a microphone to line level and then simultaneously feed it to up to 12 locations at once.



The MDA series has a balanced 150/250 ohm microphone input that will accept levels from -60 dBm to -30 dBm. The maximum output level is +21 dBm and will feed 6 balanced or 12 unbalanced 600 ohm outputs. Frequency response is +0, -1 dB from 10 Hz to 15 kHz with .03% maximum distortion. There are 3 MDA versions to choose from: a table top or a 3½" rack package each with one common level control and a 3½" rack unit with 6 individual output level controls.

Prices range from $197 to $225.

RAMKO RESEARCH, INC.
11355-A FOLSOM ROAD
RANCHO CORDOVA, CA 95670
PHONE: (916) 635-3600

for additional information circle number 71

INTERFAX ANNOUNCES GROUND FAULT TESTER

The strings on an electric guitar are connected directly to the chassis of the instrument amplifier . . . which means that the performer's body is connected to the amplifier chassis whenever he touches the strings. This fact makes the performer a "sitting duck" for any electrical current leaking out of any 110 volt operated device that he may simultaneously touch; such as a microphone, any other amplifier,

for additional information circle number 70

stage lighting hardware, or any other similar performance gear that is plugged into an outlet.

Leakage current out of a *non-defective* amplifier flowing through a human body can be as much as 150% above the threshold of sensation.

Leakage current flowing out of a *defective* amplifier, through a human being. can be as much as 10 times greater than the level considered potentially lethal.

There are cases on record of musicians who were electrocuted, and just about every veteran performer has an unforgettable story to tell about being "zapped" by a microphone.

Now a new invention by InterFax provides the first inexpensive and reliable means for testing sound system set-ups for ground faults that create shock hazards and unwanted system noises.

The InterFax Model ST2 "Electronic Safety Analyzer and Ground Fault Detector" allows the user to test any sound system set-up for the presence of leakage current that exceeds safety standards, and to also test a system for leakage current that can cause a shock and unwanted system noise. Each of the two tests is related to the level of leakage current that will flow out of the exposed metal parts of any electrical device that is pugged into a wall outlet. Systems can be tested in a matter of seconds.

The new InterFax ST2 comes with a five (5) year replacement warranty and is now available for $6.95.

INTERFAX
1008 EAST FAIRY CHASM ROAD
MILWAUKEE, WI 53217

for additional information circle number 72

EXPANDER UNIT FOR TASCAM MODEL 5'S ANNOUNCED

An expander unit, designed specifically for use with the TEAC Tascam Series Model 5 mixer, can increase the inputs from eight to as many as 20.

The extension unit – called the Model 5EX – is equipped with eight 201 input modules. An additional four inputs are available as an optional package.

According to Ken Sacks, sales manager for TEAC Tascam Series product line, the self-powered 5EX, when connected to the Model 5 mixer, retains all the functions of the Model 5, including four line output busses, a cue output buss, echo output buss and a solo output.

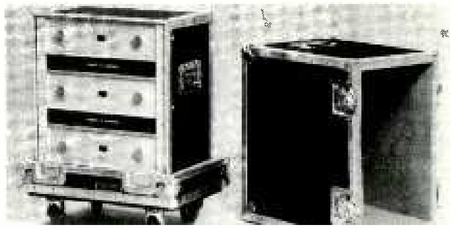
The 5EX carries a nationally advertised value of less than $1,300.00.

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

for additional information circle number 74

ANVIL DESIGNS AMP RACK TRAVEL CASE

Anvil's custom-designed ATA-type amp rack case is said to be the ultimate in protection for sensitive audio components.



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PHONE: (213) 686-1353 or 575-8614

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(continued from page 52) **INVENTORY OF SPEAKER TESTS . . .** available at a cost of $2.50 each from Speaker Tests, c/o R-e/p

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- ☐ 398 Peavy Spider 22 on CL&S RH90 with Z
- ☐ 399 Altec 291-16A on 511E with Z (0°, 30°, 45° hor.)
- ☐ 400 Emilar EA175-8 on EH800 (0°, 30°, 45°, 60° horizontal)
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- ☐ 403 Altec 802D with 23744 back radiation with Z (0°, 15°)
- ☐ 404 Altec 291-16A on 311-90
- ☐ 405 Altec 808-8A with 23744 on Emilar EH800
- ☐ 406 Altec 808-8A with 23744 on Frazier F-150 (0°, 15°, 30° horizontal)
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- ☐ 410 Peavy 1210T with Z
- ☐ 411 Altec 515B in standard box with Z
- ☐ 412 Peavy Spider 22 on University Cobraflex (0°, 30°, 45° horizontal)
- ☐ 413 Peavy Spider 22 on University Cobraflex (0°, 15°, 30° vertical)
- ☐ 414 JBL 2215 in standard box with Z
- ☐ 415 E-V DH1012 on Kustom MF-1012 with Z (0°, 30°, 45° horizontal)
- ☐ 416 E-V DH1012 on Kustom MF-1012 with Z (0°, 15°, 30° vertical)
- ☐ 417 Gauss PS-15B in standard box with Z
- ☐ 418 E-V DH1012 on CL&S RH60 with Z
- ☐ 419 Altec 808-8A with 23744 on modified E-V Sentry mid-horn (0°, 45°, 60° hor.)
- ☐ 420 Altec 808-8A with 23744 on modified E-V Sentry mid-horn (0°, 15°, 30° ver.)
- ☐ 421 Altec 808-8A with 23744 on 811B
- ☐ 422 JBL 2420 on Altec 604E horn on small baffle
- ☐ 423 Altec 807 with 23744 back radiation
- ☐ 424 JBL 2420 on Altec 604E horn without baffle
- ☐ 425 JBL 2420 back radiation
- ☐ 426 Gauss 5840 in standard box with Z
- ☐ 427 JBL 2420 on Altec 811B
- ☐ 428 E-V DH1012 on CL&S BRH90 (0°, 30°, 45° hor.)
- ☐ 429 E-V DH1012 on CL&S BRH90 (0°, 15°, 30° ver.)
- ☐ 430 4 JBL 2220 in 4 E-V 606 box (2X2 array and 4 high stack)
- ☐ 431 4 JBL 2220 in E-V 606 box (2 stacks of 2, 8 ft. apart)
- ☐ 432 JBL D130 in standard box with Z
- ☐ 434 Altec 808-16A (aluminum) on CL&S RH90 (old) with Z
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- ☐ 444 Gauss HF4000 on CL&S BRH90 (0°, 15°, 30° ver.)
- ☐ 445 Dukane 5A540 on CL&S BRH90 with Z (0°, 30°, 45° horizontal)
- ☐ 446 Dukane 5A540 on CL&S BRH90 with Z (0°, 15°, 30° vertical)
- ☐ 447 JBL 2440 on CL&S BRH90 with Z
- ☐ 448 JBL 2440 on 2355 with Z
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- ☐ 463 Frazier F°1554 in free air, measured close to cone with Z
- ☐ 464 E-V DH1012 on CL&S EC400 with Z
- ☐ 465 E-V DH1012 on CL&S EC400 (0°, 15°, 30° hor.)
- ☐ 466 E-V DH1012 on CL&S EC400 (0°, 15°, 30° ver.)
- ☐ 467 Peavy Spider 22 on CL&S EW400 with Z (0°, 15°, 30° horizontal)
- ☐ 468 Peavy Spider 22 on CL&S EW400 with Z (0°, 15°, 30° vertical)
- ☐ 469 Peavy Spider 22 on CL&S EC400 with Z
- ☐ 471 E-V DH1012 on Altec 511E with Z (0°, 30°, 45θ hor.)
- ☐ 472 E-V DH1012 on Altec 511E with Z (0°, 15°, 30° ver.)
- ☐ 474 2 JBL 2220 in Perkins box with Z
- ☐ 475 Gauss 5841 in standard box with Z
- ☐ 477 Altec 808-16A (aluminum) on CL&S SRH60 with Z (0°, 15°, 30° horizontal)
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- ☐ 485 Dukane 5P360 on CL&S BRH90 (with Z)
- ☐ 487 University ID-75 on CL&S RH90 with Z
- ☐ 488 University ID-75 on University PH Re-entrant (0°, 15°, 30°)
- ☐ 489 JBL HL180F on CL&S BRH90 with Z
- ☐ 495 Peavy Spider 22 on CL&S EW400 (with and without acoustic lens)
- ☐ 496 Peavy Spider 22 on CL&S EW400 with lens (0°, 15° 30° horizontal)
- ☐ 497 Altec 291-16B on 1003B with Z
- ☐ 498 JBL 2420 on Altec 203B with Z (0°, 15°, 30° hor.)
- ☐ 499 JBL 2420 on Altec 203B (0, 15°, 30° veritcal)
- ☐ 500 Altec 291-16B on 311-90 with Z
- ☐ 501 Altec 291-16B on CL&S BRH90
- ☐ 502 JBL 2420 on Altec 1003B (0°, 15°, 30°, 45° horizontal)
- ☐ 503 JBL 2420 on Altec 1003B (0°, 15°, 30θ vertical)
- ☐ 504 JBL 2420 on Altec 1003B (on cell axis and between cells)
- ☐ 505 JBL HL180F with D16R-2420 on CL&S BRH90 with Z
- ☐ 506 Altec 291-16B on 203B
- ☐ 507 JBL 2420 on 2350 (0°, 30° 45° horizontal)
- ☐ 508 JBL 2420 on 2350 (0°, 15° 30° vertical)
- ☐ 509 Dukane 5A440 in standard box (0°, 15°, 30°) with Z
- ☐ 510 JBL HL180F with D16R-2420 on 2350
- ☐ 511 JBL 2420 on 2340
- ☐ 512 JBL HL180F with D16R-2420 on 2340 (0°, 15°, 30° horizontal)
- ☐ 513 JBL HL180F with D16R-2420 on 2340 (0°, 15°, 30° vertical)
- ☐ 514 Dukane 5A410 (transformer in position 7) (0°, 15°, 30°, 45° horizontal with Z)
- ☐ 515 Dukane 5A410 (transformer in position 7) (0°, 15°, 30° vertical)
- ☐ 516 Altec 604C in standard box (h.f. @ max. and @ 50%) with Z
- ☐ 517 Altec 604C in standard box (h.f. @ 50%) (0°, 15°, 30°, 45° horizontal)
- ☐ 518 Altec 604C in standard box (h.f. @ 50%) (0°, 15°, 30° 45° vertical)
- ☐ 519 University ID-75-16 on RLH (0°, 15°, 30°, 45°)
- ☐ 520 JBL D130F in standard box (0°, 15°, 30°, 45°) with Z
- ☐ 521 Dukane 5A410 driver on CL&S RH90
- ☐ 522 JBL 2470 D16R2420 on Transsylvania Tube with Z
- ☐ 523 JBL 2470 D16R2420 on Transsylvania Tube (0°, 15°, 30°, 45°)
- ☐ 532 Peavy SP-1 (0°, 30°, 45° horizontal) with Z
- ☐ 536 Dukane 5P360 on 5A422 with Z
- ☐ 537 JBL 2482 on CL&S BRH90 with Z
- ☐ 538 JBL 2440 on Dukane 5A422 (0°, 15°, 30° hor.) with Z
- ☐ 539 JBL 2440 on Dukane 5A-422 (0°, 15°, 30° hor.) with Z
- ☐ 540 Dukane 5P360 on CL&S BRH90 with Z
- ☐ 541 Rauland LFS15 (0°, 45°) with Z
- ☐ 542 Emilar EC175-8 on Emilar EH800 (0°, 15°, 30°, 45° hor.) with Z
- ☐ 543 Emilar EC715-8 on Emilar EH800 (0°, 15°, 30°, 45° vert.) with Z
- ☐ 544 Emilar EA175-8 on CL&S RH90 (0°, 15°, 30°, 45° hor.) with Z
- ☐ 545 Emilar EA175-8 on CL&S RH90 (0°, 15°, 30° vert.)
- ☐ 546 Emilar EC175-8 on CL&S SRH60 (0°, 15°, 30° hor.) with Z
- ☐ 547 Emilar EC175-8 on CL&S SRH60 (0°, 15°, 30° vert.)
- ☐ 548 Emilar EA175-8 on Emilar EH800 (0°, 15°, 30°, 45° hor.) with Z
- ☐ 549 Emilar EA175-8 on Emilar EH800 (0°, 15°, 30°, 45° vert.)
- ☐ 550 Emilar EC175-8 on CL&S RH90 (0°, 15°, 30°, 45° hor.) with Z
- ☐ 551 Emilar EC175-8 on CL&S RH90 (0°, 15°, 30° vert.)
- ☐ 552 Emilar EA175-8 on CL&S SRH60 with Z
- ☐ 553 Emilar EC175-8 on CL&S ECT50 (0°, 15°, 30° toward curved side) with Z
- ☐ 554 Emilar EC175-8 on CL&S EC150 (0°, 15°, 30° toward straight sides)
- ☐ 555 Emilar EA175-8 on CL&S 1X2 (0°, 15°, 30° hor.) with Z
- ☐ 556 Emilar EA175-8 on CL&S 1X2 (0°, 15°, 30° vertical)
- ☐ 557 Emilar EC175-8 on CL&S 2X5 (0°, 15°, 30°, 45° hor.) with Z
- ☐ 558 Emilar EC175-8 on CL&S 2X5 with 2 stuffed cells (0°, 45° horizontal) with Z
- ☐ 559 Emilar EC175-8 on CL&S 2X5 (0°, 15°, 30° vertical)
- ☐ 560 Emilar EA175-8 on CL&S EC150 with Z
- ☐ 561 Emilar EC175-8 on CL&S 1X2 with Z
- ☐ 562 Emilar EA175-8 on CL&S 2X5 with Z
- ☐ 563 Rauland MLS-3 (attenuator @ 0 &@ 10) with Z
- ☐ 564 Rauland MLS-3 (attenuator @ 5) phase normal and reversed
- ☐ 565 Emilar EC175-8 on CL&S 2X4 with Z
- ☐ 566 Emilar EA175-8 on CL&S 2X4 (0°, 15°, 30°, 45° hor.)
- ☐ 567 Emilar EA175-8 on CL&S 2X4 (0°, 15°, 30° vertical)
- ☐ 568 Emilar EA175-8 on CL&S 2X4 (0°, 30°, 45°, 60° vert.)
- ☐ 569 Rauland MLS-3 (reverse phased, atten. @ 5) with and without grille cloth
- ☐ 570 Rauland MLS-3 (reverse phased, atten. @ 5) (0°, 30°, 45° horizontal)
- ☐ 571 Rauland MLS-3 (reverse phased, atten. @ 5) (0°, 15°, 30° downward)
- ☐ 572 Raulanc MLS-3 (reverse phased, atten. @ 5) (0°, 15°, 30° upward)
- ☐ 573 Rauland MLS-3 (normal phased, atten. @ 5) (0°, 15°, 30° downward)
- ☐ 574 Rauland MLS-3 (normal phased, atten. @ 5) (0°, 15°, 30° upward)
- ☐ 575 Emilar EC175-8 No. 2 on CL&S BRH90 with Z
- ☐ 576 Emilar EA175-8 No. 1 and EC175-8 No. 2 on CL&S BRH90
- ☐ 577 Emilar EA175-8 No. 1 on CL&S BRH90 with Z

EQUIPMENT

EMPLOYMENT

– BOOK REVIEW –

MICROPHONES – HOW THEY WORK & HOW TO USE THEM

This practical new book, by Martin Clifford, takes a good bit of the mystery out of microphones — as they are used in the studio, sound reinforcement, or in public address applications. It graphically demonstrates how to record almost everything from symphonies to vocal solos to singing groups, from voices with instruments to solo instruments, to small and large instrumental groups. It clearly explains how to get different sound effects and better stereo, and fully describes microphone-positioning techniques.

Well-known author Clifford discusses the many different types of microphones and accessories available, explains how to get the most out of microphone "spec" sheets, and how to interpret polar patterns. He discusses the nature of sound and the behavior of sound waves, from the basics through harmonic composition, human audibility, microphone construction, terminology and ratings, and acoustic pick-up patterns, through the specifics of acoustics in home and studio recording, public address and sound reinforcement systems. The book explains the development of a practical microphone, and details why present-day units have certain electrical characteristics, covering the many factors and design variations that affect the quality and response of various microphones. It covers the problems that will be encountered in practical recording sessions and shows how to avoid or overcome them. More importantly, Clifford tells how to make an intelligent selection of a microphone to suit one's own needs, and how to use microphones for all types of recording or amplification.

Plagued by feedback? The book's guidelines for mike selection and placement can cure it. Baffled by multi-track techniques? Frustrated by background or wind noise? Intimidated by exorbitant price tags and incomprehensible technical specifications? The solutions to these and hundreds of other sound problems are fully revealed in these nine fact-filled chapters. There is no one microphone that's best for all applications, and the most expensive may or may not be what is needed. Clifford discusses microphones from the most primitive to state-of-the-art equipment, including carbon mikes, ribbon and moving-coil mikes, and the condenser types, describing the pros and cons of each, together with their special uses. He shows how to mike any instrument, from piccolo to electric bass, and how to use microphones and room acoustics to add depth and presence without boominess, phase interference, or lyric-garbling reverberation.

This volume may be said to be two books in one, and is divided into two sections: theory and working information with emphasis on *practical* uses. Fully a third of the book is devoted to microphone applications — single- and multi-microphone applications, various monitoring schemes, and microphone placement, phasing, and recording data for over 60 different types of instruments and voice sounds. A complete chapter is devoted to miking vocals, from soloists to choirs. The special problems of the rock vocalist are discussed, with solutions for the problems of overload distortion, popping plosives (e.g., p's and t's), and selecting a microphone that will stand up under rough use. The reader will learn how a mike's position actually changes its frequency response, and how to use a mike to project intimacy or a commanding lead at the volume levels desired.

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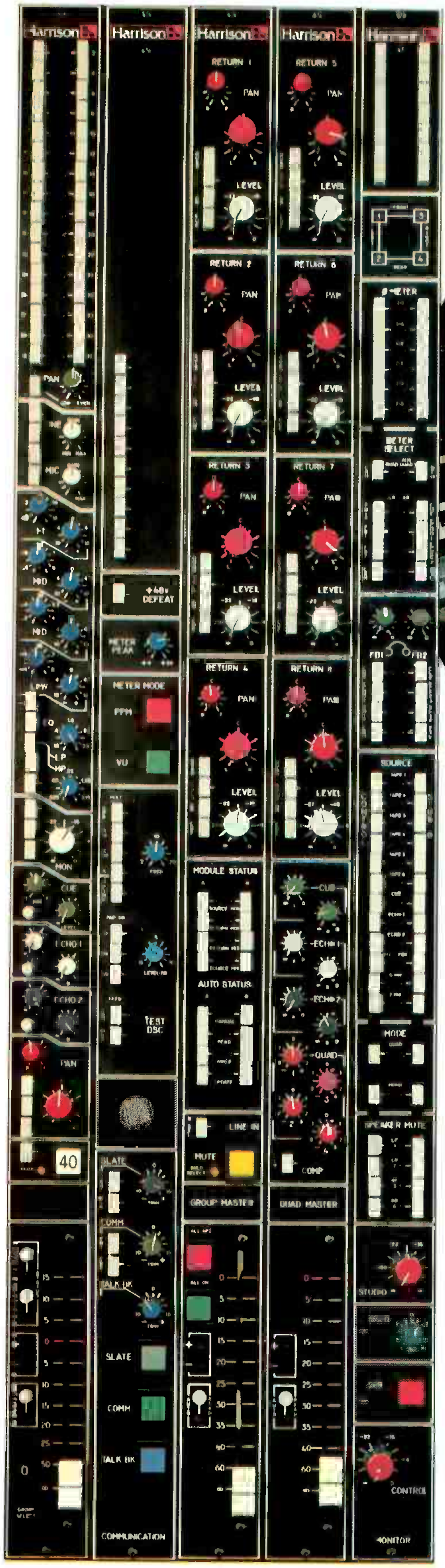

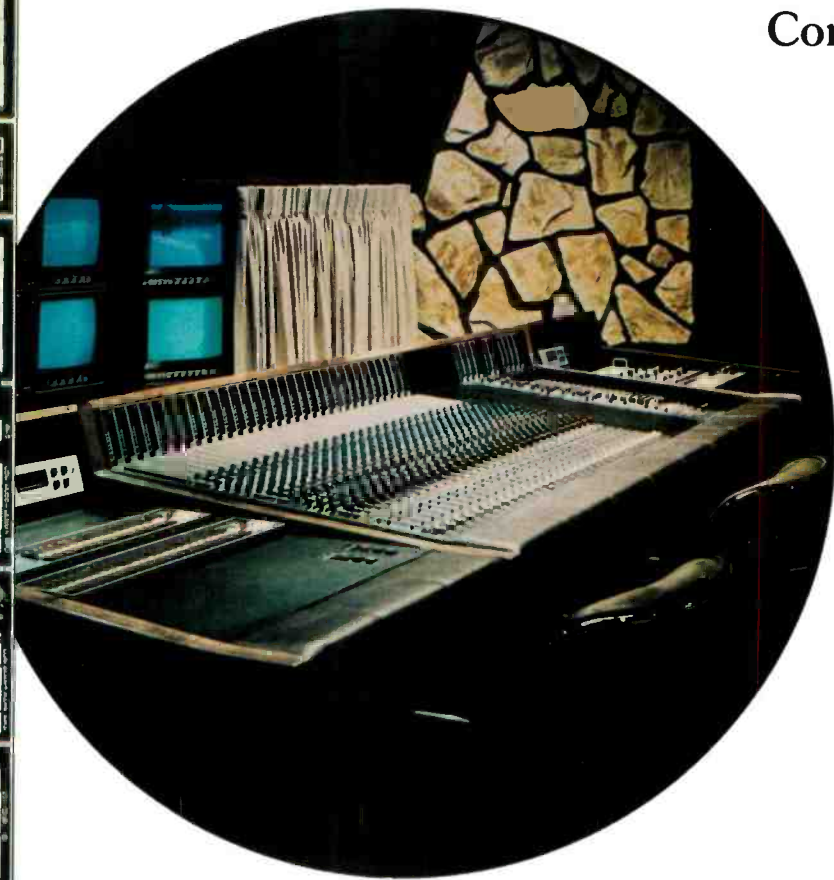

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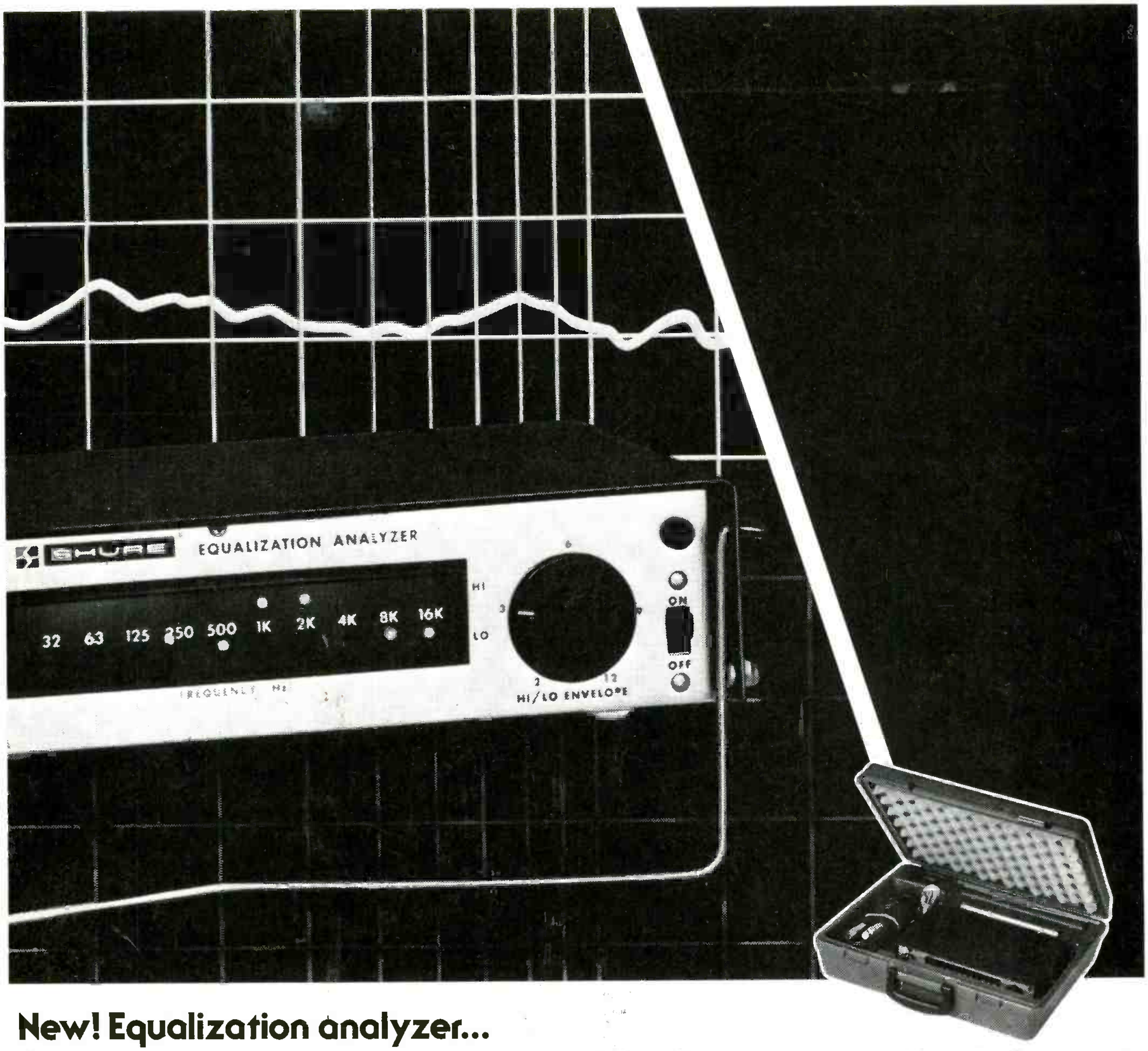

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