

P.O. Box 669, San Juan Capistrano, CA 92693 Ph: 714-728-0245 Volume 12, No. 4 Summer 1985

SYNERGETIC Working together; co-operating, co-operative

SYNERGISM

Co-operative action of discrete agencies such that the total effect is greater than the sum of the two effects taken independently.

EXCHANGE OF IDEAS

I met a man with a dollar We exchanged dollars I still had a dollar

I met a man with an idea We exchanged ideas Now we each had two ideas



THE SYN-AUD-CON

ANNUAL AWARD

FOR EXCELLENCE

IN AUDIO & ACOUSTICS

RICHARD C. HEYSER - FIRST RECIPIENT

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VOLUME 12, NUMBER 13 - WHAT REVERBERATION IS & WHAT IT IS NOT

SYN-AUD-CON ANNUAL AWARD

STREETERREPRESERE FRANKLER

VOLUME 12, NUMBER 12 - TROY MUSIC HALL

Synergetic Audio Concepts (Syn-Aud-Con) has announced the introduction of the Syn-Aud-Con annual award for excellence in audio and acoustics. The award is \$1,000 and will be given once a year to that individual who has been associated with Syn-Aud-Con either as an instructor or a "grad" and has made a significant contribution to the understanding or advancement of audio and acoustics.

The first recipient of this award is Richard C. Heyser for his development of the Heyser transform. Dick was presented this award at the Syn-Aud-Con Advanced TEF Workshop in Anaheim on August 3, 1985.

Dick Heyser's contributions to audio and acoustics include the invention of Time Delay Spectrometry, the development of the Heyser Transform, and currently a continuing series of software development for use with the Tecron TEF® Analyzer. Dick has conducted numerous special Syn-Aud-Con Workshops, written for the Syn-Aud-Con Newsletter, and addressed many Syn-Aud-Con classes on the concepts underlying the Heyser Transform.



Richard Heyser (left) - a remarkable man - sharing with class members in our Hamburg Workshop.

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DR. WOLFGANG AHNERT - "A Visitor From the East (Europe, that is)"

Dr. Wolfgang Ahnert of Berlin, GDR (American translation - East Berlin), a fascinating human being, a welleducated acoustical expert, and an enthusiastic developer of new processes, was our guest and traveling companion for two weeks this spring while we traveled from our farm in Indiana to New England and then through upstate New York to Toronto, Canada, where he again took a plane back to Europe.



Dr. Ahnert in the Boston class with Glen Ballou (left), Ted Curtin and David Ballou (a new generation of Ballou starting in audio) - those shoulders come from being a champion swimmer.

Dr. Ahnert visited Jaffe and Associates and Klepper Marshall King with us. Larry King of KMK induced the AES to host Dr. Ahnert for an evening in New York City, where he attended a Broadway show to see how they handled sound reinforcement. He then attended our Boston class and lectured on his invention, "Delta Stereophony."

Dr. Ahnert, after a stop at the Purdue Music Hall and a quick visit to the Crown factory in Elkhart, Indiana, traveled with us in our van with towed trailer through spectacular lightning and thunder storms, the Ohio and Western Pennsylvania tornadoes, and out to Glen and Debbie Ballou's in Guilford, CT. Glen and Debbie took Dr. Ahnert sailing on Long Island Sound.



Dr. Ahnert with Don and Phil Clark at Troy Savings Bank Music Hall.

Dr. Ahnert has designed and has in operation systems with as many as 380 loudspeakers in a theater, but with complete directional accuracy as to where the performer actually is positioned on the stage for every listener in the house.

After the Boston class, we spent a day with Bose seeing their remarkable manufacturing plant in Framingham, MA, called "The Mountain." Dr. Ahnert, Chris Jaffe and Wade Bray spent the day with us as guests of the Bose engineering and marketing personnel. The next day was spent measuring what may turn out to be the best concert hall in the world and certainly the best in the United States: the Troy Savings Bank Music Hall in Troy, NY (see Tech Topic Volume 12, Number 10 in this mailing). The Phil Clarks aided us in these measurements and then hosted Dr. Ahnert for an evening in their home.



Dr. Ahnert addressing the Toronto class.



Dr. Ahnert with Don and Neil Muncy.

Next we traveled to Toronto, Canada, via Niagara Falls and conducted a Canadian class. Neil Muncy, who moved to Toronto from Washington, D.C. (one glance at Toronto tells you why). helped us in our class by putting on an excellent audible demonstration of the difference between shielded cables, twisted and untwisted cables, etc., as well as a good discussion of how to solve grounding problems. Neil and his new bride had all of us as guests in their home for a truly marvelous home-cooked dinner. Neil then took over with Dr. Ahnert and was his host until his departure from Toronto.

This short recital of the two weeks from the time we met Dr. Ahnert at the Louisville, KY, airport until we said goodby at Neil Muncy's in Toronto cannot adequately picture the exchanges between two quite different political cultures but two quite similar technical backgrounds.

Dr. Ahnert worked four years in a 75' x 75' anechoic chamber with 52 sound sources and 400 subjects, studying the effects of direct sound vs early reflections plus early reflections and reverberant sound. He has the theory and mathematics. We have the TEF system. The two came together as naturally as could be imagined.

Dr. Ahnert is married and has a young daughter. He has traveled widely in the eastern bloc countries including Cuba, China, Russia and Peru. Dr. Ahnert does a great deal of acoustical work in West Germany, Austria, Japan and other western bloc nations as well. Dr. Ahnert's thorough academic background, coupled with his remarkably questioning mind, led us to an increased appreciation for contacts with the European community of acoustical authorities. You'll be hearing a great deal more about Delta Stereophony, not only from us but, we suspect, from the AES as well. In the meantime, two Syn-Aud-Con classes had a new encounter in audio. #

AN ARRAY IN DIFFICULTY

We frequently encounter a school of thought that feels that all you really have to do is get coverage. N factors, signal alignment, Q selection, all take second place to coverage. We really don't know what set of values predominated in the design of this array, but since all the horns were identical, we might guess that coverage was a main consideration.

The measurements tell their own story. Subjectively, the system users felt it had low acoustic gain, poor intelligibility, and spotty coverage. We suspect this same arrangement outdoors would be judged subjectively almost the same.



The measurement is of a single horn #7, which is near less reflecting surfaces than #3 and #5. All measurements are made in the same auditorium.

LOUDSPEAKER ARRAY CONFIGURATION H.F. 1 Figure # 1 H.F. 2 H.F. 3 H.F.4 L.E. H.F. 5 H.F. 6 H.F. 7 H.F.8 Figure # 4 $F_R = 250 \text{ Hz}$ $F_R = 250 \text{ Hz}$ Linear 0 - 10,000 Hz Linear 0 - 10,000 Hz TECRON TEP TECRON TEF 10 High Frequency Horn #5. High Frequency Horns #3 and #5.



SMILE

There comes a time in the history of every project when it becomes necessary to shoot the engineers and start production.

From Bob Hagenbach

SYN-AUD-CON NEWSLETTER SUMMER 1985

VIENNA LOVES DOGS



At a table next to us in Vienna, a lady held her dog on her lap. Loving dogs also, I asked if I could take her picture. (I was carrying my camera to take pictures during measurements at Musik Verienssaal.) Before I could take a second shot for backup, the waiter brought the dog a luscious bone, signaling his approval of the dog in his restaurant. #

BASIC INSTRUMENTATION

Perhaps the most frequent question asked after a basic Syn-Aud-Con class is what instruments should I buy. The list is simple and not really expensive. (It's far more expensive to be without the necessary tools.)

The Basic List

- 1. A high impedance, high sensitivity AC voltmeter with a frequency response at least as great as the audio range.
- 2. An impedance meter or bridge. It is important that this instrument accept an *external* oscillator. It is helpful when its internal oscillator has multiple frequencies available as well.
- 3. An oscilloscope. Frequency range 500 KHz, sensitivity at least 1mv/div. Calibrated sweep and amplitude are essential. It should be small, lightweight and easily transportable.
- 4. An audio oscillator with a *known, fixed* output impedance. Frequency range minimum 20 to 20,000 Hz. Output level at least 3 volts open circuit.
- 5. A 1/3 octave real time analyzer with a built in pink noise generator and a reasonable quality microphone.

These five basic instruments should not cost you much more than \$4,000. for first class quality, accuracy and reliability. If you are skilled in their use, no further instrumentation is required.

Desirable Extras

If the budget allows, then the following instruments are extremely useful and the money they cost is easily recoverable from typical small audio jobs, just from *the time* (real labor) they save:

1. A precision sound level meter.

2. A level recorder system.

Example Products

These products are not specially recommended or thought to be the only ones that can do the job at a reasonable price. They do meet the requirements listed above.

- 1. H.P. 3466A multimeter.
- 2. The CVS 400 impedance meter by Electro-Dynamics.
- 3. Any H.P. or Tektronix oscilloscope.
- 4. The Loftech oscillator, TS-1.
- 5. The Crown RTA.
- 6. The GenRad 1933 sound level meter.
- 7. Neutrik 3300 level recorder system.

Total price of all equipment is less than \$10,000.00.

We will be developing here at Syn-Aud-Con a series of measurement practices relevant to the use of these seven instruments by sound contractors. #

INTEGRATION TIME OF THE EAR

The Peutz disk as programed by Jebelian at Tecron allows a choice of "integration times" in the running integration of an ETC display, both the percentage of total power between two cursors as well as the calculated RT_{60} between the same two cursors. The question then naturally arises, "What are meaningful integration times?"

Shown in the figure by Per Bruel are the time constants for the human hearing system, which indicate that from 35 to 40 msec would be useful. The 1/f time for the lowest frequency of interest to us is

$$\frac{1}{20 \text{ Hz}} = 50 \text{ msec}$$

Schroeder feels that the first 15 dB of decay in a 2.0 sec room is of interest, i.e., 500 msec.

The smallest signal delay difference we found that Syn-Aud-Con classes could detect (over a single loudspeaker where two inputs are separated by an introduced signal delay) was 20 microseconds (20 usecs). Finally, we know that the Haas effect triggers from 15 to 30 msec with 20 msec being very typical of rooms we actually work in.



Relation between loudness level and duration time. (After Munson.)



Schematic drawing of the human ear with the teletransmission system's most important time constants, and the transmission characteristics of the outer and middle ear.

It's on the basis of the above considerations that we usually run two different integrations--one at 20 msecs and a second at 40 msecs. This is not to infer that other times shouldn't be used in special cases or in different views of regular cases. What we are inferring is that either 20 or 40 msec represents an excellent starting point for initial investigations with this new disk. Our experiences to date are that it is one of the most potent architectural acoustic tools we've ever had the privilege to use, especially when viewed by one who started such measurements thirty plus years ago with large wave analyzers coupled by chain drive to heavy level recorders just to get a single frequency reading. #

WHAT IS "GAIN" ?

Gain is the term used to describe the change in *level* expressed in decibels at the listener's ears upon the insertion of a device into the system in place of a piece of wire. #

THEATRE AUDIO CONTROL SYSTEM



On rare occasions, Syn-Aud-Con feels a product is sufficiently advanced that it should be brought to the general attention of our grads. Lynn McCroskey's TAC-86 Theatre Audio Control System is such a product. We have seen and heard these units in operation and feel that they could have wide application in many systems.

Pictured from top to bottom are the Sonics TAC-86QP Que Panel and TAC-86 Theater Audio Controller.

For further information, contact: SONICS ASSOCIATES, INC., 237 Oxmoor Circle, Birmingham, AL 35209. #

TIME & FREQUENCY VIEWED BY A SLM

The following was sent to us by an anonymous contributor and we felt it had sufficient interest to be shared in the Newsletter. It is a very clear explanation of why sound level meters have both peak and impulse options on them today. The discovery that speech is a complex waveform is often followed by later recognition that it, more often than not, is fairly represented by the ANSI curve on a random noise generator.

One reason that the "B" and "C" weighting networks did not give the expected results is that the equal loudness contours were based on experiments with pure tones -- and most common sounds are not pure tones, but very complex signals.

When more detailed information about a complex signal is required, the frequency range from 20 Hz to 20 kHz can be broken up into sections either one octave or one third octave wide. This is done by electronic filters which reject all signals of frequencies outside the selected band. For example, an octave filter with a center frequency of 1 kHz permits sounds in the 707 to 1410 Hz range to be measured, but rejects all others. This process where a signal is analyzed in many frequency bands is termed frequency analysis. The results are presented on a chart called a spectrogram.

If a sound is of short duration, that is, less than one second, it is termed an impulsive sound. Practical examples are typewriter and hammering noise. This represents another problem in loudness evaluation because the shorter the sound, the less sensitive the ear will be in perceiving it. Researchers generally agree that for sounds shorter than 70 milliseconds (70 thousandths of one second), the perceived loudness decreases. This has resulted in agreement on a standardized electrical circuit whose sensitivity decreases with short duration sounds. This circuit is called the "Impulse" characteristic. However, the damage risk is not necessarily reduced although the loudness decreases with short signals.



For this reason, some sound level meters include a circuit for measuring the peak value of the signal, independent of the signal's duration.

BASIC MATH CLASS

All complex concepts are constructed from basic building blocks. The most sophisticated computer program consists of a mass of sub-routines, each of which should be straightforward and of itself easy to understand. Invariably, in audio, we encounter sub-routines that in themselves are valid, but when brought into conjunction with another sub-routine are found to be invalid. A good example is the use of the marvelous Sabinian based equations in non-reverberant spaces.

We have been doing a great deal of rethinking about audio and acoustic basics in recent months as we construct the 1985 Syn-Aud-Con basic seminars. We can't help but wonder if there's an interest in learning about basic mathematics of use in audio. We know there's a need, but that doesn't mean that those in need have an equal interest in doing something about it.

We'd like to hear how many of you would be interested in a basic audio and acoustics math class that would start with Ohms law and go forward through algebra and geometry review, into a basic conceptual view of Laplace, Fourier, Hilbert, and Heyser transforms. Attendance would require that you bring your own H.P. electronic calculator. If enough of you have an interest, we'll plan such a workshop in our future schedule. #

AN EXCEPTIONAL OPPORTUNITY

Robert Haworth, Audio Engineer for the Reorganized Church of Jesus Christ of Latter Day Saints has a working Western Electric 25B console in excellent shape. Price is not a major consideration, but he would like to find a home for it where it would be kept in its original condition for the use and edification of future generations. The quality of this console is what made early FM sound as good as it did and, hopefully, someone of the Syn-Aud-Con family knows a museum or school that would treasure such an artifact.

CONTACT: Robert Haworth, RLDS Church, P. O. Box 1059, Independence, MO. 64051. #



Western Electric

Will Parry Maryland Sound 4900 Wetheredsville Baltimore, MD 21207 (301) 448-1400

Eric Hofbeck-Boeing Noise Tech Lab Boeing Commercial Airplane Co. P. O. Box 3707 MS/1W-03 Seattle, WA 98124 (206) 655-5429

HHB Hire & Sales Unit F-New Crescent Works Nicoll Road, London N2109AX England

NEW TEF OWNERS

Allan Peerson Peerson Audio, Inc. 511 S. Olive Avenue West Palm Beach, FL 33401 (305) 832-1921

Platinum Marketing 75 Renfield Crescent Whitby, Ontario L1P 1B3 Canada

Martin Audio 5456 Stanhope St. Euston, London NW13EX England Woody Smith Cyber-Kinetik, Inc. 11839 Starcrest Drive San Antonio, TX 78247

D. Feldman NOS Post Box 10 Hilversum, Holland 120008

Leatham Electronics 58 Kent Terrace Wellington New Zealand

> SYN-AUD-CON NEWSLETTER SUMMER 1985

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SCHEDULE 1985

Seminars

Workshops

CHICAGO, IL Ramada-The O'Hare Inn September 26-27

CLEVELAND, OH Cleveland Airport Marriott October 1-2

NEW YORK AREA The Hilton at Harmon Meadow October 10-11

WASHINGTON, D.C. AREA Ramada at Tyson's Corner October 22-23

November 5 - 7 (2 1/2 Days) \$600 Nashville, TN

> Staff: DR. EUGENE PATRONIS, Georgia Tech CLIFFORD A. HENRICKSEN, Electro-Voice, Inc. EDWARD M. LONG, E. M. Long Associates

FUNDAMENTALS OF LOUDSPEAKER DESIGN

INSTALLING & TROUBLESHOOTING THE SOUND SYSTEM

November 18 - 20 (3 Days) \$600

Atlantic City, NJ

(On Location in a Casino Show Room)

FARREL BECKER, Audio Artistry

DAVID KLEPPER, KMK Associates

PHIL CLARK, Diversified Concepts Inc.

CLEARWATER, FL Ramada Inn Central October 30-31 "On Location" - November 1

> ATLANTA, GA Perimeter North Inn November 13-14

Conventions

Staff:

AES - OCTOBER 13 - 16, NEW YORK, NY

V.M.A. Peutz of Peutz and Associates, Ron McKay of BBN (LA office), and Don Davis will conduct a three hour Workshop at the New York AES on Concert Hall Acoustics and Measurements.

Hospitality Suite -

We will hold a 2 Day Hospitality Suite at the AES Convention. Check with our sponsors' exhibits for Suite #.

ASA - NOVEMBER 4, NASHVILLE, TN

Dr. Eugene Patronis and Don Davis will give a paper in Russ Berger's session on Recording Studio Acoustics. The paper is entitled, "Reverberation Time in Physically Small Rooms." #

SMILE

PEER'S LAW: The solution to a problem changes the nature of the problem.

VOLUME 12, NUMBER 4

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"PRINCIPLES AND APPLICATIONS OF ROOM ACOUSTICS"

The two volume set entitled *Principles and Applications of Room Acoustics* by L. Cremer and H. Muller (translated by T. J. Schultz) is expensive (\$94.25 for Volume I and \$69.75 for Volume II) and expansive in its total coverage of European work of fundamental value, and exceptional in terms of its instant use in equipping a TEF user to jump eons ahead of those without these references. How to *really use* the new integration capabilities available for the ETC measurements are brought forth with clarity, detail, and accompanied by mathematical data of great value.

European work on the "fine structure" of reverberation and their contributions to the psychoacoustics of Concert Halls is a vast storehouse of literature beautifully summarized and categorized in these two volumes. My volumes now have so many markers placed in them that they look like porcupines.

Ted Schultz's translation and his careful, restrained use of correlating notes calling attention to similar work in the U.S.A. when truly applicable enhances the value of these volumes even more than the original.

We happen to be in a "Golden Age" of acoustical measurement capability and our most pressing problems are "what should be measured" rather than, "I wish I had a way to measure it."

The Europeans understand the opportunity present as they have already discovered that:

"The measurements (directional behavior of sound in a large auditorium) make it clear that the assumptions underlying the statistical theory are <u>very poorly fulfilled</u> in a large hall, even in the later time period, and *not at all* in the early period."

TEF users have at hand the apparatus to shape the needed new theories. The authors of these two volumes have outlined the best starting points, ths most likely directions, and a majority of the false traits.

Syn-Aud-Con feels these books are so vital that we are willing to serve as a collection point for orders which could then be placed as a group order for, hopefully, a price break. If you are interested, contact us before October 1, 1985.

CREMER, L. and H. MULLER (Translated by T.J. Shultz.) Principles and Applications of Room Acoustics, Volumes 1 & 2 1982. vol. 1; 660p. \$94.25. 0-85334-113-3. vol. 2; 456 p. \$69.75. 0-85334-114-1.

MARKETING IN THE BARNYARD

Don was writing the Newsletter at the farm in Indiana and had in front of him the data on combining speakers in a cluster!! For about a half hour, he stared out in space and shuffled the papers. Finally, I asked him what was bothering him and he said he had something on his mind and he might as well write it first to clear his brain. - Carolyn

Marketing In The Barnyard

There are the swine, wallowing in their expense accounts; the sheep and goats, waiting for their job evaluation; the cattle, munching their security. Over here's the banty rooster who thinks he's an eagle and spends a great deal of time crowing over other hen's eggs. The horse is a noble creature with one end engineering and the other end the sales manager. Naturally, the compost pile belongs to the marketing manager, as that is what they use for brains, what the customer thinks of their product, and what their plans and forecasts are made of. #

OPUS 3

In our last Newsletter, Vol. 12, No. 2, we wrote about Opus 3 records from Sweden. We received a phone call from Alan Kanter telling us that the distributor in the United States was in nearby San Clemente. The records are available from Scandinavian Sounds, P. O. Box 3656, San Clemente, CA 92672. Phone 714-498-0709. You may purchase records or CDs. The price is \$18.

The records are exceptional. For a start, purchase Test Record 1 (Depth of Image), catalog No. 7900 and No. 7900CD; Test Record 2 (Timbre), catalog No. 8000; and Test Record 3 (Dynamics), catalog No. 8300. They will become your reference test records. The Opus 3 test discs are actually samplers from their entire catalog. They will test your music system. We have been using the new Spica loudspeaker that Bert Whyte praised so highly a few months ago. We knew how carefully John Bau has worked with his TEF® analyzer, being one of the very first TEF analyzers shipped. John tells us that the Spicas are being used as nearfield monitors in recording studios. It's easy to see why. #

SYN-AUD-CON AND THE FOURIER TRANSFORM

We were surprised recently to realize that our strong stand against the use of inadequate FFT analyzers and the misuse of high quality FFT analyzers for two port acoustic measurements has been largely misinterpreted to be an attack on the Fourier Transform itself.

The Fourier Transform, i.e., either a discrete or fast Fourier Transform version as is the case in all present day devices, is used in the TEF analyzer to allow various global to local transformations of data *already acquired by not using the Fourier Transform*. The TEF analyzer utilizes the "modulation domain" and the "signal delay" domain from the general case Heyser transform in the actual acquisition of data. The Fourier Transform is a mathematically degenerate special case (a hyperplane) within the Heyser transform. The Heyser transform, (not at present fully utilized by any analyzer but so far the most developed in the TEF analyzer) is a hyper surface, i.e., curvature can be accommodated The kernal of the Heyser transform.

While we presently use the Heyser transform to go from one domain with its dimensionality to another domain with the same dimensionality (just as does the Fourier transform) we go through other domains on the way that do not have the same dimensionality as our beginning and ending domains.

Those of us in audio are not equipped at the present time to utilize new domains of altered dimensionality any more than Fourier's contemporaries were ready to build an FFT analyzer. We can be sure, however, that FHT (fast Heyser transform) analyzers will spring from the present TEF analyzer before the turn of the century. Many of us are thrilled to be conscious witnesses to such a revolution in human thinking.

Conclusion

Syn-Aud-Con advocates the proper use of the Fourier transform and all of its special algorithms. We do not advocate *FFT Analyzers* as ideal or efficient for two port acoustic measurements. We sincerely hope that all of you will help make this distinction clear to the uninformed. #



NOTE....

The rolls of film from the Dallas and Kansas City classes are missing. If we locate, we will put in the next Newsletter. #

EPICYCLES

Many of you in recent classes have heard us compare the epicycles (whorls or loops with loops) on the TEF® Nyquist display with the epicycles of the Ptolemic theories (see the figure at right for a typical Nyquist type display of an epicycle).

The view reproduced here clearly reveals that a careful astronomical observer would indeed believe in epicycles (I saw them with my own eyes). The age old conflict has been described by Fred Hoyle -- "Nothing pleases the observer more than disproving the theoretician, and nothing worries him more than discovering something thoroughly new, since this opens up new fields of activity for the theoretician....



The apparent motions of the planets on the sky, simulated in a planetarium. (Courtesy of Munich Planetarium.)

Actually, the work of both observer and theoretician is necessary to progress in science. Without one, the other would soon stultify."

The Hoyle book, Astronomy and Cosmology, published by W. H. Freeman and Company, San Francisco, has, in the fourth section on the solar system, an excellent chapter entitled "The Pioneers of Astronomy." Mr. Hoyle is as sure of his hypothesis as any of the ancients. While a splendid reporter of the past, he remains dogmatic about the present, apparently unable or unwilling to look at his "certainties" as if they might just be as certain as the "certainties" of the past. #

SMALL ROOM RESONANCES



Many of us have listened in small control rooms to the low frequency resonances that occur when one of the dimensions of the room supports a particular frequency like a "tuned" tube. Figure 1 is such a resonance (about 125 Hz). Figure 2 is the same measurement made after the construction of a Helmholtz resonator (by Doug Jones). #

QUOTE FROM GEORGE SZELL

"In music, one must think with the heart and feel with the brain."

SHURE FP16 DISTRIBUTION AMPLIFIER

Shure has started down the road to a series of truly useful "field production" products. The latest is an audio distribution amplifier, the FP16. It's lightweight (6 lbs. 1 oz.) with one input and six outputs. It comes with a complete instruction manual and, except for using "phase" instead of "polarity" and some other minor semantic differences, is a very useful guide.

Inputs



	IMPEDANCE (at 1 kHz)		INPUT CLIPPING		IMPEDANCE (at 1 kHz)			
	FOR USE WITH	ACTUAL	LEVEL AT 1 kHz	OUTPUT	FOR USE WITH	ACTUAL	LEVEL AT 1 kHz	
Mic	150 ohms	1k	-62 to -6 dBV*	Mic	150 ohms	0.5 ohms	- 34 dBV	
Line	Less than 10k	66k	- 12 to + 44 dBV*	Line	600 ohms	180 ohms	+ 16 dBV	
Link	Less than 5k	24k	+8 dBV	Link	600 ohms or	100 ohms	+ 16 dBV	
*Depende	ent on input control	setting			greater	or less		

Dependent on input control setting.

There are several features this distribution amplifier has that particularly appeal to us:

- 1. Transformer coupled inputs and outputs.
- Individual input and six output "gain" controls. (Shure continues to use the term "voltage gain" which lies on our ears with the same jar as an obscenity.)
- 3. There are:
 - (A) Mic level outputs
 - (B) Line level outputs
- (C) Link circuit outputs (to allow more than one D.A. without sacrificing regular outputs).
- 4. Battery and A.C. operation (A.C. for 120 and 240 V A.C.).

After our recent Grounding and Shielding Workshop, I am a real advocate of transformer coupled battery operated devices for use in the field. One of the best ways to isolate from the AC ground is to run the system from batteries. Ideally, future systems may well plug into AC only to keep batteries charged and all operating current will be taken from the batteries.

We now have from Shure a series of components that are our first choice for small high quality sound systems. There are quite a number of systems where the Shure M 267, the FP 16 and, hopefully, a pending rebirth of their excellent 105A amplifier in packaging appropriate to professional use would be a hard-to-beat "package." #

COMB FILTER "STOP BANDS"

The first notch of a fundamental frequency f_1 is found by: $f_1 = \frac{1}{4}$

where: f_1 is the first notch

t is the signal's delay time in secs.

Example

Two loudspeakers separated by two inches will have an f_1 of:

 $f_{1} = \frac{1130}{1/6} = 6780 \text{ Hz}$ $f_{2} = 2 \frac{1130}{1/6} = 13,560 \text{ Hz}$ $f_{3} = 3 \frac{1130}{1/6} = 20,340 \text{ Hz}$ $f_{4} \cdot \cdot \cdot \cdot$ d = tc

where: d is the misalignment in feet or meters, c is the velocity of sound in ft/sec or M/sec.

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A BIT OF NONSENSE

We are always interested to read in audio periodicals discussions between manufacturing personnel ("We've never had that problem before."), consultants ("God (i.e., the architect) told me to do it."), and the sound contractor ("I'm always low bid."). We love them all but not all the time. In any case, this "ideal" bit of "boiler plate" arrived on our desk from Russ Berger of Joiner-Pelton-Rose, Dallas, and the only way we know of to dispose of it is to publish it.

SUPLEMENARY KONDISHUNS TOO ALL CONTRAKTORS

The work we want did is clearly showed on the attached plans and specerfications. Our enjineer, whose had plenty of college, spent one hell of a lot of time when he drawed up these here plans and specerfications. But not nobody can think of everything. Once your bid is in, that's it bruther! From then on, anything wanted by our enjineer or any of his friends, or anybody else (cept the contraktor) shall be considered as showed, specerfied, or emplied and shall be pervided by the contraktor without no expens to nobody cept the contraktor.

If the work is done without no extry expens to the contraktor, then the work will to tooken down and did over agin until the extry expens to the contraktor is satisfactory to our enjineer.

Our enjineer's plans is right as drawed. If drawed wrong it should be discuvered by the contraktor, kerrected, and did right at his own expens. It won't cut no ice with us nor our enjineer if you point out mistakes, which our enjineer drawed. If you do, it will be one hell of a time before the contraktor does any more work with us or him.

The contraktor is not sposed to make fun of our enjineer, his plans, or the kind of work we're doin. If he do, it's jist too bad fer him.

Any contraktor walkin aroun the job with a smile on his face is subject to revue of his bid.

If the contraktor don't find all our enjineer's mistakes before he bids this here job, or if he ain't got enuf sense to know that our enjinear is going to think up a bunch of new stuff that got to be did before the job is completely did, then it's jist too bad fer him (meaning the contraktor).

The contraktor gotta use all good stuff on this here job -- none of that crap from (area of your choice).



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EARTHS, GROUNDS, SHIELDS ...

We recently purchased a small (144 page) expensive (\$21.95) book by Ralph Morrison, entitled *Instrumentation* Fundamentals and Applications, publisher Wiley Interscience, ISBN 0 471 88181-3.

This is an excellent text for any technician serious about learning the engineering of earths, grounds (read common) shields, balanced and unbalanced circuits, transformers, differential inputs, RF interference and a host of other details necessary to a pure signal transmission accomplished without compromising personnel safety.

Tidbits, such as, Time delay (read signal delay) is a function of filter cutoff frequency, filter type, and filter order. Further on, "The phase slope $d\phi/df$, where ϕ is the phase angle and f is the frequency, is most affected by the least-damped poles in the filter. An adjustment on this one damping factor can serve to match time (read signal) delays between channels."

That the book is practical is found in the statement, "Instrumentation must operate on the assumption that there will be ground potential differences between all signal connection points" and later, "The signal leads that connect to a ground should not be labeled 'earth' or 'ground' because building codes may force a safety treatment. Labels, such as, signal return, signal common, and guard can be used to avoid this problem."

Mr. Morrison is a straight-to-the-point, straight-from-the-shoulder, straight-shooter. Syn-Aud-Con recommends his book as a useful self tutorial text. #

FIELD TESTING OF A TOA MCX-106

From David Dunaway, Audio Department, Hardin County Materials Center in Elizabethtown, KY:

It's been a while since we last spoke. It was in Chicago at the September '84 seminar. I really enjoyed the class. It made me realize how much I <u>didn't</u> know about audio. Since the class, I have evaluated the sound systems in all of our schools and have either made improvements or started on projects to improve the quality. The results have been very rewarding. Thank you both.

Now for the real reason I'm writing. I've noticed in your newsletters you frequently talk about a product. If I may add my two cents, I would like to relate a situation that happened to us.

We recently purchased a TOA self powered mixer, model MCX-106. It's a very fine instrument for sound reinforcement. We use it mainly for our board meeting room and occasionally use it out on the "road" to run stage monitors. Well, it was one of



TOA MCX-106 Powered Mixer

those occasions that has caused me to appreciate the quality of engineering that TOA invests in their product.

After a performance at one of our schools recently, we loaded our equipment into our sixteen foot stepvan equipped with shelves. The TOA was placed on the top shelf where it usually rides. Now in case you're not familiar with this part of the world, Kentucky roads are not the best maintained. On our journey back to the office, we went over some pretty mean stretches of road. So rough in fact, it caused the shelf on which the TOA was sitting to collapse. The poor thing fell an estimated 5 1/2 feet (to the floor of the truck). I panicked. We stopped the truck. 'Oh my God,' I thought, 'it will never work again!' I picked it up and the only visible damage was one corner of it was mashed down slightly. Once we arrived at the office, I rushed it into the shop and quickly started connecting speakers and a microphone, praying all the while. Well, to our surprise, it worked! Really!! Right down to the equalizer and built in cassette deck. I'm being conservative when I say it took a'hell of a lick.'

It's not often I am so impressed with a product that I'll take the time to write about it. Please pass this information on to the TOA people. They have a good product if it can stand up to our abuse. (Oh! I almost forgot, the TOA left a 2 1/2" by 3" dent in the floor of the truck. Poor truck.)

Again, thank you for your wealth of knowledge and help. I'll be looking forward to your upcoming seminars and workshops. #

AUDIO & ACOUSTICS BASIC CLUB

Ken Wahrenbrock has issued Volume 1, No. 1 Newsletter (April) for the new computer/calculator club he has started. Ken knows how many calls we get asking, "Where can I get audio programs for my calculator/computer?"

Presently he has programs for the IBM-PC, Apple II, and Commodore 64. Ken has valuable, menu driven programs and is looking for more.

Write Ken for more information on the club: 9609 Cheddar Street, Downey, CA 90240. #



IMPULSE PARAMETERS

As an impulse approaches, the ideal of a Dirac impulse (Dirac delta function) of zero time and infinite amplitude the bandwidth of the signal in the frequency domain becomes wider, following Bw = $\frac{1}{T}$.

Thus, an impulse with a T = 50 usecs becomes a bandwidth of 20KHz $\frac{1}{0.00005}$ = 20,000.

Seemingly, this signal would be an ideal source if it were not for a fatal flaw in energy accounting. As the time interval shortens, the amplitude must increase in order to maintain the same energy level. By the time a useful time interval is achieved in terms of a useful bandwidth being generated, the signal's amplitude is so high as to overdrive almost any test device. If the amplitude is reduced, so is the total energy content and the resultant signal-to-noise situation becomes so severe that meaningful measurements are not possible in any kind of normal "in situ" environment. In artifically quiet environments, impulse measurements can be instructive tutorial examples of sources for Fourier transformation of signals. (Compared to swept sine wave techniques, impulse signals suffer roughly an 86dB disadvantage.)

Cross Spectrum Parameters

The Cross Correlation function between the input and the output of a device excited by a white noise signal source is identical to the impulse response of that device. This allows the test signal to be continuous, thus increasing the energy contained in the test signal and allows the use of a test signal with a known flat frequency response. The maximum available power from this technique falls about 10dB below that for swept sine waves due to the difference between the crest factors for each type of signal. The use of dual channel FFTs does provide a benefit in S/N over an impulse signal for a greatly increased price and complexity.

Manfred Schroeder has suggested the use of "Galois sequences" also known as maximum-length sequences or pseudorandum shift register sequences, as ideal sources for precision measurements with extremely low energies in the presence of strong interfering noises. (Cross spectrum measurements with dual channel FFTs.) Schroeder uses such signals to measure sound systems with the audience present, but with the *test signal inaudible to them.* He uses averaging times in excess of one hour. Even better S/N can be realized with TEF using either scalar or vector averaging for an averaging time of a minute. #

MODULATION TRANSFER FUNCTION

The concept of the Modulation Transfer Function (MTF) for the measurement of speech intelligibility has received a great deal of attention from various sources including an instrumentation manufacturer hoping to have it imposed as a standard, thereby providing a need for an expensive instrument to measure it.

We don't disagree that MTF *properly* employed can serve as an indicator of reduced signal transmission of information. We do feel that MTF is a very poor predictor at the drawing board stage. In spite of the Journal blitz currently going on, we still feel that Peutz's approach wherein he uses L_D, L_R, L_{AMB} , and RT_{60} is the most reliable measurement. Insofar as classic statistical parameters have any effect, Peutz's %AL_{CONS} equation is still more than acceptable.

The real problem in both measuring and predicting speech intelligibility of sound systems in auditoria is the fact that most intelligility problems have their roots in source misalignment, early reflections (within one foot of the source), misequalized drivers, severe crossover network anomalies, bad microphones and microphone placement, mismatches in critical circuits containing passive devices, and perhaps most prominent of all, poor coverage of the audience areas.

Only an academician or an instrument manufacturer, hungry for a new marketplace, could seriously propose MTF as a useful working tool likely to help solve any of these fundamental causes of unacceptable intelligibility. #

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COMMENTS ON ECAN

From Topper Sowden, Principal at Joiner-Pelton-Rose, Dallas, Texas:

I compliment Herbert T. Chaudiere on his Tech Topic regarding electronically controlled ambient noise (ECAN) Tech Topic Vol. 12, No. 4. His description of the function of these systems and their perils and pitfalls regarding construction has been proven many times to my firm.

I would like to share some thoughts regarding statements mady by Mr. Chaudiere in his article.

With respect to Mr. Chaudiere's comments regarding dual-channel systems, we have seen system failures occur within six months after the system had been turned on. The use of a redundant system did assist the client in getting through the time period while awaiting replacement of the component (in one case, it was an equalizer). The output cable of the damaged equalizer was relocated to the output of the other equalizer to allow full distribution of masking sound throughout the facility during repair of the equalizer. I agree that this is a more expensive approach to masking, but the major expense of a masking system involves the provision and installation of the loudspeakers.

This technique can go too far, as we are aware of some designs involving two-way (low frequency and high frequency) loudspeaker systems with redundancy on the high frequency transducers. We have found this type of system to be more expensive than the value received by the client after its installation.

The redundancy of systems and hop-scotch wiring of loudspeakers in the ceiling does provide a reasonable gain with further diffusion of the sound field in the open plan. This is actually demonstrated every time we adjust a two-channel system. Our equalization process involves adjustment of all loudspeakers in an area by single equalizers before reconnection as a two-channel system. We, therefore, observe the system in a single-channel mode before modifying the system for two-channel redundant operation.

Again, my compliments to Mr. Chaudiere on a truly enjoyable article. As a consulting firm with over 15 million square feet of masking floor space, we can tell that Mr. Chaudiere has 'seen it before.' #



LOUDSPEAKER Q AND DIRECTIVITY INDEX DI

The directivity factor Q can also be expressed as a directivity index D₁. The relationship is:

$$D_{I} = 10 \text{ LOG } Q$$
 and $Q = \frac{\begin{pmatrix} D_{I} \\ 10 \end{pmatrix}}{10}$

From the above it can be seen that D_I is a level and Q is a power ratio.

In psychoacoustic tests involving varying Qs, it is usually easier to evaluate the likely subjective effects of changing Qs by looking at them as $D_{\rm I}s$.

-	Q	DI	Q	DI	Q	DI	-
	1	0	12	10.79	40	16.02	
	2	3.01	13	11.14	50	16.99	
	3	4.77	14	11.46	60	17.78	
	4	6.02	15	11.76	70	18.45	:
	. 5	6.99	16	12.04	80	19.03	:
	6	7.78	17	12.30	90	19.54	
	7	8.45	18	12.55	100	20.00	
	8	9.03	19	12.79	•	•	,
	9	9.54	20	13.01	•	•	
	10	10.00	30	14.77	•	•	
1.1	11	10.41			260	24.15	1

Note that for equally audible steps, if you started with a Q = 1, the next choice of Q = 4 would be followed by 16, 60, and 260. To make audible changes in Q from a lower to a higher value usually requires quadrupling the Q or halving the DI distance, both of which accomplish the same thing, a 6 dB increase in the direct-to-reflected ratio.

Remembering that 3.01 dB is, to a majority of listeners, a just noticeable difference, and that 10 dB is considered twice as loud for mid-band sounds, the experienced sound engineer makes adjustments in his parameters, when called for, that are clearly addible to an untrained listener.

When in a truly reverberant space, i.e., 2 KHz octave band in excess of 4.5 secs, high Q devices are mandatory.

Most intelligibility problems spring from array misalignment, very early reflections from surfaces near loudspeakers, failure to achieve sufficient acoustic gain to allow safe operation below the feedback level, thereby amplifying the room's reverberation through the sound system, and the off axis response of the loudspeaker, all are important factors. #

AN ALTERNATIVE dB ADDITION TECHNIQUE

Gary Berner in the Kansas City class reminded us of an alternative decibel addition equation. What was interesting to us was that Gary had discovered this for himself in one of his college classes.

Example

If I wish to add 90 dB to 96 dB, I take the difference in dB (6 dB) and do the following:

10 LOG $\begin{pmatrix} & \frac{-6}{10} \\ 10 & +1 \end{pmatrix}$ +96 = 96.97 dB

SHARING

I met a man with a dollar. We exchanged dollars.

Now I have only 72¢. The other guy was a Canadian.

Told to us by Robert Davis, National Film Board of Canada. Attributable to Michael Drolet.

#

COMMUNITY M-4 DRIVERS

Community M-4 drivers are quoted as producing 100 acoustic watts. That's an Lw of:

Lw = 10 LOG
$$\left(\frac{100w}{10^{-12}w}\right)$$
 = 140 dE

That's 140 dB at 0.283M for a Q = 1

 $\frac{0.283M}{1} \qquad \frac{100 \text{ cm}}{1 \text{ M}} \qquad \frac{1 \text{ in}}{254 \text{ cm}} \qquad \frac{1\text{ F}}{12^{\text{m}}} = 0.928^{10}$

Therefore, we could make the following table:

<u>M-4</u>	level at 10'	Lp at 100'	Q
	119.36	99.36	1
	129.36	109.36	10
	132.37	112.37	20
	134.13	113.13	30
	135.38	115.38	40
	136.35	116.35	50
	137.14	117.14	60
	137.81	117.81	70

M-4s were installed on eight Community PC horns which were hung from a large crane parked on a barge and towed out into the river for a good coverage point for the audience along the shore.

The sound system covered several thousand people on the shoreline who witnessed the APBA Formula 1 World Championship Hydroplane Race.

Community Light and Sound, Inc., is located at 333 East Fifth Street, Chester, PA 19013.#



SIGNAL MISALIGNMENT AND CROSSOVER NETWORKS

Suppose you have a crossover frequency of 800 Hz. The key quesion to ask yourself is what misalignment will cause a notch filter to coincide with the -3 dB crossover point with the resulting octave wide hole in the total response.

$$t = \frac{1}{F_1}$$

Where t = the misalignment time in secs.

 f_1 = the crossover frequency in Hz.

Therefore: $t = \frac{1}{800Hz} = 0.00125$

and the misalignment distance would be: d = tc

Where: c = is the velocity of sound in ft. per sec or m/sec

d = is the misalignment distance in feet or meters.

Therefore: 1130(0.00125) = 1.41'

Unfortunately, many real life devices, such as, midrange horns and woofers, end up with this kind of distance separation and when it actually exactly coincides with the crossover's -3dB frequency, really sizeable holes appear in the total frequency response. #

VOLUME 12, NUMBER 4

ANSWER TO GROUNDING & SHIELDING EXERCISE



TI-35II - A GENUINE BARGAIN

In this day and age, bargains are not that frequent; but we ran across one in a Kansas City Montgomery Ward's store.

The TI-35II Scientific Calculator does every calculation anyone is ever likely to encounter in audio, has a constant memory and comes with a really useful 224 page textbook written by the staff of the Texas Instruments Learning Center in cooperation with the staff of the University of Denver Mathematics Laboratory plus many contributions from TI users. As if that's not enough, there's a 48 page quick reference guide that slips into the wallet sized carrying case.

Keys include $\frac{1}{2}$, $\frac{1}{2}$

FLOOR MICROPHONE MOUNTING, 1966



Pete showed up at our Grounding Workshop in Minneapolis in May with photos of the real thing along with Paul Veneklasen's original drawings (which carry the date 4 March 1966. #

Delay line eases comb-filter design

by Hanan Kupferman Century Data Systems Inc., Anaheim, Calif.

Designing a multiple-frequency notch filter with standard band-stop transfer functions is complicated and tedious because the process needs repeating. In addition, filter matching is difficult for sections with different



The September 22, 1982, issue of *Electronics* magazine, pages 154-155, contained this interesting use of these ideas, reproduced here in partial form. Note particularly that subsequent notches appear at $2/T_D$, $3/T_D$, etc., and are automatically set at the harmonics of the chosen fundamental frequency. This is one of the benefits obtained in the PZM® system by placing the microphone element on a bar above the "pressure" surface, thus enabling the We wrote about the "Historical Forerunner of the PZM" in the Newsletter, V12 N1, page 14. Jake Ewalt said Pete Alward from Portland mentioned using Mr. Veneklasen's floor mounts. "Maybe he can get photos of the real thing."



band-stop frequencies. Through exploiting the characteristics of a delay line, this comb filter is easily realized and provides a band-stop response for a fundamental frequency and its harmonics. The fundamental frequency corresponds to the delay time of the delay line.

The general scheme of the filter (a) shows that the delay line with a delay τ and characteristic impedance R is driven by and equals matched impedance sources R_1 and R_2 . Use of input and output buffers in the circuit is optional. The fundamental frequency where the first notch occurs is $f'_1 = 1/\tau$. Other band stops occur at $f_2 = 2/\tau$, $f_3 = 3/\tau$, and so on.



tuning of any desired frequency for a series of such notches (current units have this first notch out around 70,000 Hz or a spacing of approximately 16/1000ths of an inch). These relationships are so fundamental and so simple that they have escaped the attention of a number of academicians pontificating on the subject. #

BOOK REVIEW

We recently came across a most useful new mathematics book entitled *Handbook of Electronics Calculations for Engineers and Technicians* edited by Milton Kaufman and Arthur H. Seidman, publisher - McGraw Hill. A great deal of the newer digital jargon is covered after a thorough basic set of chapters at the front of the book.



Shown here, slightly modified, is an illustration from the basics chapter. It would be difficult to sum up the centuries of mathematical development more succinctly.

We recommend this volume to anyone seeking to keep up-to-date on mathematical notations, techniques, and applications in electronic engineering. \blacklozenge



SYN-AUD-CON NEWSLETTER SUMMER 1985

CLASSIFIED

FOR SALE: Western Electric 25B Speech Input Console. See Page 8 for details. CONTACT: Bob Haworth, RLDS Church, P. O. Box 1059, Independence, MO 64051.

JOB OPPORTUNITY:

Coordinate the setting up and operation of professional projection equipment and sound reinforcement systems in support of instructional activities for 22 large lecture halls on campus. Coordinates the setting up and operation of AV equipment, multiple format videotape playback and TV camera, multi-media and sound reinforcement systems in support of University conferences and programs. Trains, assigns, and schedules approximately 20 student assistants to handle workloads.

<u>QUALIFICATIONS:</u> Two year Technical Degree (preferably in Audio Engineering with one year or more experience). Three years experience in planning and set-up of complex audio systems, in trouble-shooting and analysis of infield operational failures and providing preventative maintenance on professional projection equipment and audio systems. Proven oral communications skills.

CONTACT: Sinette Winfield, Manager of Employment, Personnel Department, SUNY/Buffalo 106 Crofts Hall, Buffalo, New York 14260.

the ultimate put-down

Bruce Thayer of Interconnect in Cedar Rapids sent this marvelous "put-down" from Science News:

Your excellent coverage of the New Orleans symposium honoring P.A.M.Dirac (SN: 6/20/81, pp 394, 396, 397) brings to mind an anecdote related to me years ago by Prof. Ernest Pollard, who directed my Ph.D. work in nuclear physics at Yale during the late 1930s. Pollard was a graduate of Cambridge and studied under Lord Rutherford and James Chadwick.

It seemed that after one of Dirac's rare seminar lectures to the graduate students at Cambridge the moderator stepped to the podium and said Prof. Dirac would be happy to entertain questions. Immediately a hand went up in the audience and the individual stated in a loud voice, "Prof. Dirac, I must say I don't understand Equation 4."

Everyone sat quietly anticipating a response from the eminent man, but none came. Dirac remained motionless in his chair staring straight ahead. After a full minute had passed, the moderator became noticeably nervous and inquired, "Prof. Dirac, would you care to answer the gentleman's question?"

"That was no question," replied Dirac dryly. "That was a simple statement of fact."

#

SMILE

Read A. Wineland of Audio Marketing Associates, Brecksville, OH, reports the following news story from NBC's "Today Show":

Brooke Shields, while working on a Bob Hope Television Special, narrowly escaped electrocution when she dove into a pool of water forgetting she still had her 'wireless microphone' on. "She was very lucky," observers said.

Read comments, "Maybe OSHA should outlaw those dangerous 9 volt batteries." #

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Senn

NEUTRIK



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Syn-Aud-Con receives tangible support from the audio industry, and eleven manufacturing firms presently help underwrite the expense of providing sound engineering seminars. Such support makes it possible to provide the very latest in audio technology while maintaining reasonable prices relative to today's economy and to provide all the materials and continuing support to all graduates of Syn-Aud-Con.

Personnel from these manufacturers receive Syn-Aud-Con training which provides still another link in the communications circuit between the ultimate user and the designer-manufacturer of audio equipment. They are "in-tune" with what a Syn-Aud-Con graduate needs.

Their presence on this list as a Syn-Aud-Con sponsor indicates their desire to work cooperatively with you in professional sound.

> Bose Corporation Community Light & Sound, Inc. Crown International, Inc. Emilar Corporation HM Electronics, Inc. Industrial Research Products, Inc. JBL Incorporated/UREI Electronics Neutrik Products Shure Brothers Inc. Sunn Electronics Switchcraft, Inc. TOA Electronics, Inc.

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