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The Editor's Corner

PSYCHOACOUSTIC PHENOMENON

TRY to write a simple sentence with your eyes closed. You will then appreciate that to perform such an action easily and accurately, you must see what you are doing. Similarly, in speaking or in playing a musical instrument, you must hear the results. It is difficult to shut out the sound because of bone conduction in the head; however, for experimental purposes, we once tried a continuous artificial noise to mask our voice. When speaking under these conditions, we had a peculiar feeling of not remembering from word to word what we were saying. We were not sure whether we had just said a connected sentence, or whether it was merely an unvoiced thought.

Accidentally, during the early 1940's, we discovered something that went a step further. We were monitoring a wire recording as it was being made, with a pickup head spaced a fraction of a second after the recording head. If the announcer wore the monitoring headphones, he immediately became speechless. What he had to cope with was not silence or random noise, but the same intelligence normally fed back to his hearing, except that now it was hopelessly out of phase. The effect was so unexpected and so overwhelming that a few small wagers were won from the uninitiated who thought they could recite a short paragraph while wearing the magic headphones. One could beat the machine by blurting out brief phrases intermittently, with a pause after each burst to collect one's thoughts, or by speaking very slowly. We made ground rules to cover these situations.

Studies were made later by others, in a more quantitative fashion, with control over time delay, volume, and other factors, so that subjective differences would be noticed. It is an interesting field for psychoacoustic study, and much remains to be done. There are differences among individuals in the degree of confusion or frustration generated by delayed listening. Women seem less susceptible than men, which supports the theory that some of the more voluble talkers never got into the habit of listening even to themselves.

We have the makings here of a testing machine which might separate the sensitive from the "thick skinned" and which could evaluate one's abilities as an orator, a salesman, or a politician.

—MARVIN CAMRAS, *Editor*

NEGATIVE FEEDBACK PHILOSOPHY

The plate resistor feedback scheme mentioned in the editorial of the January-February, 1959, issue, was described in an RCA Application Note in 1938 or 1939. I have used it many times since with excellent results. An article published somewhere or other objected to this method on the grounds that negative feedback from the plate of a pentode output stage will increase the hum level. Ripple voltage in a poorly-filtered "B" supply will not create much 60-cycle current ripple through the output transformer primary if the plate resistance of the pentode is high; but it will if plate resistance is made low by feedback. The article may have discouraged potential users from trying this circuit, even though it was excellent in actual operation.

Feedback from the voice coil does not have any bad side effects, imaginary or otherwise. The only circuit problem here is oscillation. It yields readily to the treatment described in the 1940 edition of Terman. Yet many of the older (and less old) "Hi-Fi" amplifiers operated near the edge of ultrasonic of ULF oscillation. To the industry, the problem seemed difficult. The audio industry (referring to consumer equipment manufacturers) has seemed, like the broadcast receiver business, to be attuned to a very low grade of engineering. I think that much of the good circuit development work in these fields has been "bootlegged," *i.e.*, devised by engineers on their own initiative, without the direction or support of management.

It appears from these TRANSACTIONS that a considerable part of our technical progress in this field still comes from amateurs—defining an amateur broadly as one who is not being paid to do the specific thing he is doing.

I am inclined to believe that as an engineering activity becomes more specialized (*e.g.*, the design of home broadcast receivers), it develops a network of empirical rules and taboos in which it finally becomes enmeshed. The process is furthered by the risk attending an unproved design feature in a mass-produced article. The final result is a sort of paralysis. An improvement that is actually simple becomes, to the specialist, a fearsome and gigantic departure. For twenty-five years, radio sets have been manufactured which can be made to sound better, for example by putting in an output transformer weighing a pound more, by adding a little feedback, or by sticking a little padding in the cabinet. But to the manufacturer and his "work-horse" specialist in set design, the expense was unthinkable—although money was readily spent on cabinet trim, clocks, remote-control devices, and dummy speakers.

Doubts as to consumer desires were dispelled by surveys which showed that the public wanted what it was already getting. I do not think that these surveys and jury listening tests were rigged; but the temptation to tell a client what he wants to hear is very strong. A parallel case is the recent use of Motivational Research by the automobile industry.

The TV set industry is in most cases the same as the home radio set industry of prior years, and a direct heir to its attitudes. So long as a profit can be made by following the ancient traditions, why risk a change?

Third Person Passive

On thinking about the third person passive, I believe that it is practical and frequent practice to rig it a little.

Example, straight third-person passive:

> Noise was found in the voltmeter circuit. The source was traced to the cathode bypass capacitors, which were found to be leaky. These components were replaced by higher-voltage units.

Rigged:

> Leaky cathode bypass capacitors produced noise in the voltmeter circuit. Higher-voltage units effected a cure.

The actual nature of the process is not entirely clear to me on such short consideration, but it appears to consist of assigning active roles to inanimate objects.

LAWRENCE FLEMING
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Bellaire, Tex..

PGA News

CHAPTER NEWS

Albuquerque—Los Alamos

Frank Rodocy, Chief Engineer of Audio Devices, spoke on "Tape Technology" on January 8, 1959.

Don Davis and Paul Klipsch of Klipsch and Associates told about their "Journey into Hi-Fi—Brussels Fair Equipment Demonstration" at a meeting on February 27, 1959.

Cincinnati

"The New McIntosh Stereo-Preamplifier" was described by Frank McIntosh at the January 20, 1959 meeting.

On February 11, PGA members were guests of King Records, with Harold Neely as master of ceremonies.

John S. Boyers of Bell Sound Systems presented a paper on "The Mechanical Aspects of Magnetic Recording" on March 17, 1959.

Dayton

An informal listening session of stereo tape, courtesy of Frank C. Morrison, was held on February 19, 1959.

"Design Parameters of Stereo Phono Cartridges" by Robert C. Avelon of Electro-Voice was featured at the March 19 meeting.

For the April 2 meeting, Robert P. Palomo of Ohio Bell Telephone Company spoke on "New Techniques and Devices for Telephony."

A tour of Baldwin's factory at Cincinnati, Ohio was conducted by John Quitter of the Baldwin Piano Company on April 16, 1959.

San Diego

Glidden Barstow of the Naval Electronics Laboratory reported on the Los Angeles Audio Fair papers at the meeting held on February 25, 1959.

San Francisco

Lawrence W. Johnson, in the April San Francisco *Grid*, writes:

> Jack Petrak of the Hewlett-Packard Co. was speaker at the March meeting of the Professional Group on Audio. The meeting, held in the Hewlett-Packard engineering conference room, had as its subject the first public discussion and showing of a new harmonic-wave analyzer, the -hp- Model 302A.
>
> The basic circuitry for any wave analyzer was first discussed, and the similarity to the common broadcast band superheterodyne receiver was pointed out. Basic differences are found in the mixer, which must be balanced; in the IF, which usually utilizes a crystal filter; and in the fact that the output presentation is on a meter.
>
> Previous harmonic-wave analyzers have generally been large heavy instruments, rather awesome in appearance, and fairly complicated to set up and operate. Among the objectives which guided the several-year-long development of the 302A were the following:
>
> 1) The instrument was to be transistorized.
> 2) Sharper selectivity was desired.
> 3) An automatic frequency-control system should be provided.
> 4) It should be much easier to use than its predecessors. There should be no trap-door adjustments, and such features as a simplified attenuator presentation, reduced size and weight, linear frequency scale, and a high input impedance should be provided.
> 5) Use as a tuned voltmeter should be facilitated by providing a test signal source whose frequency tracks the frequency to which the meter is tuned.
> 6) It should make available an output signal at the frequency of the component being measured.
>
> All of these objectives were accomplished as a result of a sizeable group effort. Some of them represented basic design decisions about the course to follow in developing the instrument, while others were departures from previous standard techniques. Transistorization automatically brought size and weight under control; and in minimizing warm-up drift problems also reduced the need for trap-door setup adjustments. Sharper selectivity was a natural goal, and led to the inclusion of an afc circuit. In turn, the afc provides the desirable ability to stay tuned in on a component even though the source frequency may drift.
>
> Crystal filters are used in the IF amplifier and afc discriminator. As a result, the selectivity curve is approximately 50 db down, twenty cycles away from center frequency.
>
> Two special features involving output signals are worthy of note. In normal and afc operation, an output signal is available at the frequency of the component being measured. This is accomplished by heterodyning the IF output with the local oscillator and rejecting components over 50 kc. Such an output is useful in determining the harmonic frequency exactly by means of an electronic counter.
>
> The other special feature works in similar fashion when the instrument is used as a tuned voltmeter, as for transmission measurements. The heterodyning and filtering described above are used, but the afc's crystal discriminator is converted to a crystal oscillator whose output beats against the local oscillator. The result is thus a constant-amplitude signal of the same frequency as that to which the instrument is tuned, available at a front panel jack. Use of this bfo feature eliminates the need for an external signal source when making transmission measurements through noisy or distorting networks.
>
> The instrument, normally powered from the ac line by a rectifier-filter-regulator, can also be battery-operated. In this condition the regulator is still used, so that a wide range of battery voltage variation can be tolerated without degrading the wave analyzer operation.
>
> The meeting concluded with a discussion and demonstration of the instrument's capabilities. Measurements were made of the harmonic components of a 1000-cycle square wave—up to the 49th—and of the characteristics of a low-pass filter, the latter using the bfo mode of operation.

Twin Cities

Robert L. Sell, Chairman 1959–60, reports that:

> The initial meeting of the newly formed Twin Cities Chapter of the Professional Group on Audio was held on the evening of March 18, 1959. Those present numbered twenty-three. We were quite pleased with the turnout, considering that there are thirty-six local members of the Professional Group.
>
> Officers were elected as follows:
>
> | | |
> |---|---|
> | Chairman | Robert L. Sell |
> | Vice-Chairman | Richard F. Dubbe |
> | Secretary-Treasurer | Joseph F. Dundovic |
>
> The newly elected officers met after the meeting to organize the initial activities of the Chapter. Plans were laid for a plant tour of a local recording studio as the next event on the calendar. A speaker was also decided upon for the joint session meeting to be held in June; lastly, a third chapter meeting was planned. We feel that these three meetings will bring us up to the summer season and give us a rousing start.—J. Ross MacDonald

ANNOUNCEMENTS

WESCON

Audio papers scheduled for presentation at WESCON, San Francisco, Calif., on Tuesday afternoon, August 18, 1959, are:

1) "A Resonance-Vocoder and Base-Band Complement: An Hybrid System for Speech Transmission," by J. L. Flanagan of Bell Telephone Labs.
2) "Novel Compression-Expansion Method for Audio and Video Use," by W. R. Aiken of Ross Radio Corp. and C. Susskind at the University of California, Berkeley.
3) "A New Stereophonic Projection Console," by B. B. Bauer and G. W. Sioles of CBS Labs.

This year's program features a "Panel of Peers" made up of experts in the field who comment on the subject matter of the papers presented during the session.

NEC

An audio program at the National Electronics Conference to be held October 12–14, 1959, at the Sherman Hotel in Chicago, was organized in cooperation with PGA by Bill Ihde. Papers scheduled are:

1) "Free Field and Pressure Calibration of Microphones by the Reciprocity Method," R. W. Benson, Armour Research Foundation, Chicago.
2) "A Transistorized Stereo Preamplifier and Tone Control for Magnetic Cartridges," A. B. Bereskin, University of Cincinnati, Ohio.
3) "A Compatible Tape Cartridge," M. Camras, Armour Research Foundation, Chicago.
4) "Hiss Reduction in Master Tape Machines," A. A. Goldberg, CBS Labs., Stamford, Conn.
5) "Design and Use of RC Parallel-T Networks," G. White, White Instrument Labs., Austin, Tex.

Wide-Stage Stereo*

PAUL W. KLIPSCH†

Summary—**Stereophonic playback systems are studied to determine the accuracy with which they can reproduce the geometry of the original sound. Sounds were generated in a geometric pattern, recorded, and reproduced over a loudspeaker array. The methods used were similar to those used by Steinberg and Snow in 1933. Using two sound tracks, a derived center playback channel, and corner placement of flanking speakers, geometry plots were made with almost as good accuracy as when observers listened to an actual person speaking at the indicated stations in the geometric array.**

Since wide speaker spacing was used, corner placement becomes natural.

An evaluation of corner speaker placement from the tonal standpoint (as contrasted with the geometric) shows that there is a large increase in quality available by taking advantage of the reflections of the floor and walls. It is shown that corner placement of flanking speakers and use of a derived-channel center speaker affords the best reproduction of geometry as well as tonality.

INTRODUCTION

IN 1933, personnel of the Bell Telephone Laboratories performed what is believed to be the first demonstration of stereophonic sound reproduction using loudspeakers.

When I was invited to demonstrate the 1958 version of stereo before the Phonograph Club and Hi Fi Hobbyists of the Bell Telephone Laboratories, I accepted with humility, for I realized the great debt the present art owes to the Bell Telephone Laboratories, and to what extent present workers have been standing on the shoulders of giants.

Thus as I bring coals to Newcastle, I'd start by mentioning the work of Steinberg and Snow in creating the idea of three-channel stereo using two sound tracks. That they dismissed it lightly may be due (in my private opinion) to two causes; first, their main effort was on three independent channels, and second, there might have been an unvoiced feeling that this derived channel consisted of something for nothing, like perpetual motion.

Perhaps my own contribution would be limited to suggesting an underlying philosophy aimed to establish the derived channel as a real one. It can be stated thus: "If two microphones are properly placed relative to each other and to the sound source, their combined output is that of a single microphone in the middle. The output of this microphone that wasn't there may be recovered by recombination. The output of a physical third microphone can also be recovered."

The idea is analogous, though not similar, to the phantom circuit used in wire communication.

HISTORICAL

Stereophonic sound reproduction stems largely or perhaps entirely from the experiments and demonstrations conducted at Bell Telephone Laboratories in 1933.[1]

With the development of tape recording, auditory perspective took on vitality, with the result that the lay public is asking which is better, "high fidelity" or "stereo."

Stereo was demonstrated at the Chicago Parts Show in 1952 with a stage that was too narrow or the stereo angle too small to realize appreciable improvement over "monophonic" reproduction.[2]

At other shows, two speakers were typically some 15 feet apart and the stereo angle less than 20° for most of the listening area. To this error of insufficient angle was added the close spacing of microphones approaching binaural separation. Many comments have been heard that the observers witnessed no "stereo" effects.

A demonstration at the Los Angeles Audio Fair in 1953 employed a wider angle, but the effect was that of three peep holes rather than a curtain of sound. It is suspected that the events were too closely spaced to the respective microphones, resulting in the effect of three isolated sound sources.

EXPERIMENTS

To determine the minimum and optimum stereo angles, some experiments were tried. Two speakers were set up outdoors, 7 feet apart. One observer detected the stereo effect at a distance of over 50 feet. Expressed in angles, the minimum stereo angle (angle subtended at the listener by the distance between speakers) ranged from 9° to 16° for the various observers, with an average of 12° for all observers. The source was a two-track tape recorded with two microphones suspended 15 feet high, 26 feet apart, and about 6 feet forward of the front row of musicians; the group was a 60-piece band.

The preferred angle for the test conditions was from

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* Manuscript received by the PGA, January 23, 1959. Revised manuscript received, May 15, 1959. Presented before the Phonograph Club and Hi Fi Hobbyists of the Bell Telephone Laboratories, Arnold Auditorium and West Street Auditorium, New York, N. Y., November 21 and 24, 1958.

† Klipsch and Assocs., Hope, Ark.

[1] Steinberg and Snow, "Symposium on auditory perspective," *Elect. Engrg.*, vol. 53, pp. 9–32, 214–219; January, 1934.

[2] The term "monophonic" was suggested by W. B. Snow, one of the contributors to the symposium on Auditory Perspective. He preferred this term over the misnomer "monaural," which means one-eared.

19° to 70°. The average for all observers was 37°, corresponding to a ratio of distance of about 1.5 to 1.

Indoors, in a room 16 by 25 feet with about typical living-room acoustics, the stereo effect was lost at a listener distance about three times the speaker separation for all observers and twice the distance for most observers.

Stereo effect is here used to mean merely that there is ability to distinguish sounds originating from the right and left sides. In the next section, using wider angles, reproduction of stereo geometry will be studied, and "stereo effect" will be accorded quantitative meaning. The same source material was used as for the outdoor test. The room was treated to simulate the liveness of an "average" living room.

The angular minima were easily determined. The preferred distances varied widely with the individual listener.

Indoors, only one of eight observers could detect the stereo effect at 24 feet with speakers separated 7 feet. At a distance of 14 feet, a speaker spacing of 14 feet was preferred to 7 feet. The basis of preference was merely how the observer liked the sound of a symphonic band, without regard to the accuracy with which the original geometry was duplicated.

The usually-preferred geometry was comparable to the angle subtended by a listener at a distance of 40 feet from a musical group spread out on a 40 foot array. It is believed the angle is independent of the scale.

REPRODUCTION OF STEREO GEOMETRY

When one arrives at a desirably wide stereo angle, there sometimes appears a separation of events.

A recording was made of a vocal announcement at each of eight stations in front of two microphones outdoors. Fig. 1(a) shows the arrangement. With wide separation of two speakers on playback, the events appeared clustered at the speakers. The same wide speaker separation with a derived third channel added[3] resulted in a curtain of sound and a plotting of original sound locations about as accurate as the plots obtained with the three-channel stereo described by Steinberg and Snow in 1934. Fig. 1(b) shows one of the individual plots selected as being about "average" of the six plots made by three observers. Fig. 1(c) shows a plot made "live."[4] Fig. 1(d) shows a plot using only two-channel playback with wide spacing.

In this playback array, the room was the same as in other tests; the speakers were 25 feet apart and the observers on array axis at a distance of 15 feet from the array. For three-channel playback, two plots were made by observers 10 feet off axis. These were only slightly less accurate than plots made by observers on axis.

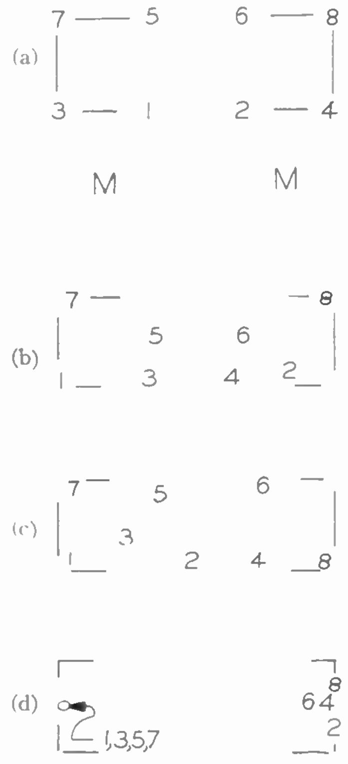

Fig. 1—(a) Array of sound source locations and microphones, located outdoors. (b) Sound generated as in (a) was recorded and then played back over a "wide-screen" stereo array. This is one of the plots of one observer. (c) Plot by the same observer of the original sound. (d) Plot by the same observer using two-channel playback.

A test tape of a symphony orchestra with soprano soloist was also used. This was recorded with two microphones approximately 22 feet apart, 15 feet above and 15 feet in front of the performers. The soloist stood some 6 feet to the left of the podium. On playback she seemed to be about the same angular distance to left of the center speaker, and the orchestra filled the space between flanking speakers. More recent listening tests with tapes made with three microphones indicate excellent geometry, but not having been present at the recording session the author can make no statement as to "accuracy of geometry of reproduction."

SCALE EFFECTS

Ideally, the sound-source array of Fig. 1(a) would be reproduced exactly in the stereo playback, but with this difference; the sound array would be contained within the speaker array in spite of the fact that its extent is greater than the microphone array. Fig. 1(b)-(d) bears this out.

If the array of Fig. 1(a) is 24 feet wide with 16-foot microphone spacing, and the playback geometry uses a 24-foot speaker array, the scale is 1:1. If one recorded an orchestra with 25-foot microphone spacing, the scale would be about $1\frac{1}{2}$:1, but the angles subtended by the listener would remain the same and the orchestra would be heard in approximately the original geometry and at some distance behind the wall against which the speakers are arrayed.

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[3] P. W. Klipsch, "Stereophonic sound with two tracks, three channels by means of a phantom circuit (2PH3)," *J. Aud. Engrg. Soc.*, vol. 6, pp. 118–123; April, 1958.

[4] This is an actual plot made by an observer wearing a hood and attempting to locate the voice of an actual person speaking from the indicated station without intervening microphones or speakers.

THE CORNER EFFECT

At least one large producer of stereo records and tapes recommends corner speaker placement. It has been recognized for at least a quarter of a century that corner placement improves speaker response. To evaluate this fact quantitatively, a non-corner speaker was set up and tested in three environments, first on legs about 14 inches high and with the speaker system about 4 feet from the room walls, next on the floor at the same plan location, and lastly in the corner. Fig. 2 shows the response obtained in the three locations, curve 2.1 being on legs, curve 2.2 on the floor, and curve 2.3 in the corner.

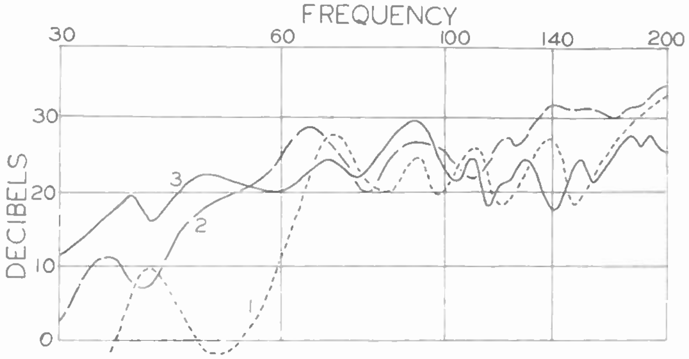

Fig. 2—Advantage of corner speaker placement is shown by the smoother and more extended response for speaker located close to wall surfaces. Curve 1, speaker on 14-inch legs, about 4 feet from walls. Curve 2, speaker on floor, 4 feet from walls. Curve 3, speaker in the corner.

TWO-CHANNEL STEREO

While the only true stereo playback system must be conceded to be three-channel, it was felt that various two-channel configurations should be studied.

From a geometry standpoint, a pair of small portable speakers was chosen. These were three-way, woofer, midrange, and tweeter, but with restricted bass to permit portability.

In Fig. 3(a), the stereo effect was discernible on the axis of symmetry back to about twice the distance between speakers.[5] In Fig. 3(b), the stereo effect was evident at an actual greater distance due to the wider separation, and also over a greater distance from the line of symmetry. In Fig. 3(c), the stereo effect was at its best with the widest stage and largest listening area. Fig. 3(d) obeyed the law of maximum stereo distance, about twice the speaker spacing, resulting in a very small stereo listening area. Fig. 3(e) proved, as expected, to provide a wide-stage stereo, by wall reflection, but the frequency response anomalies of wall reflections were undesirable.

In all but Fig. 3(e), the stage was bracketed by the speakers, that is, no sound appeared to come from outside the space between the speakers. In Fig. 3(e), the stage was room width, but this configuration displayed the biggest hole in the middle, and also showed severe errors of frequency response. Fig. 3(c), as expected, afforded the best frequency response with about an octave more bass range.

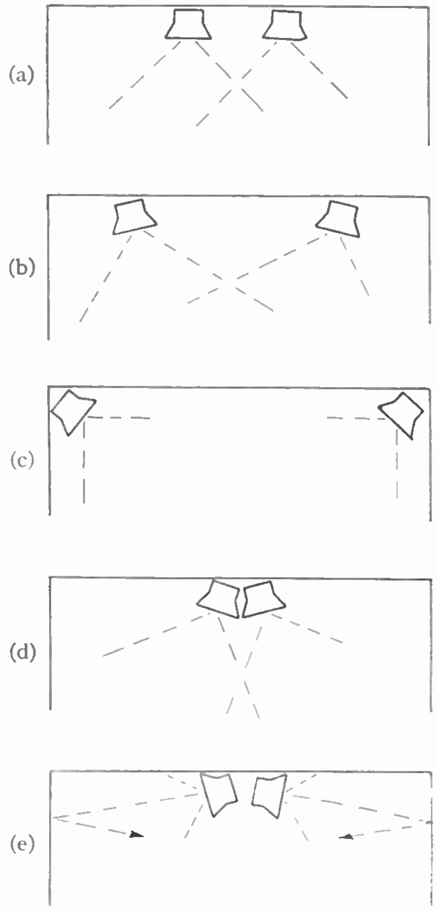

Fig. 3—Various configurations tried for two-channel stereo. Stereo geometry improved progressively from (a) to (c). Configuration (d) displayed negligible stereo effect. Configuration (e) resulted in a wide-screen stereo effect, but the frequency-selective reflections resulted in echo-like highs and a severe hole-in-the-middle effect.

In all the configurations, Fig.3 (a)–(e) the test tape based on Fig. 1(a) gave results showing principally two-point sources. On a musical program, a soloist could be focused for a listener on axis, but movement of the listener caused the soloist to defocus in one direction or the other, the least objectionable array being that of Fig. 3(c), and the most objectionable that of Fig. 3(e).

Tests depicted in Fig. 3(a)–(c) serve to confirm the necessity of the third channel. This agrees with the findings of Steinberg and Snow.[1] They found that observers in various parts of a listening area localize a given sound source at different virtual positions, and that the use of the third channel reduced the differences in localization for various observing positions.

Along-the-wall placement is usually illustrated as in Fig. 3(a). Comparison of the configurations shown in Fig. 3(a)–(c) shows that all exhibit a stereo effect. The maximum distance at which the stereo effect may be heard is proportional to the stage width, and therefore the listening area is approximately proportional to the square of the array width. For a single observer at opti-

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[5] Same 16×25-foot room with speakers on the short wall.

mum distance, there seems to be little choice between the three configurations except, of course, as the corner placement adds to the tonal range.

All three configurations tended to give a two-point source effect, pointing up the necessity for the third channel.

Conclusion

1) The third channel added to a narrow along-the-wall array results in an improved curtain of sound within the narrow confine of the speaker spacing.

2) The next step in improvement is made by widening the array. In a room with a dimension ratio of 3 to 2, the suggested limit of array width is the long wall of the room.

3) When the limiting locations are used, diagonal placement in the corner offers improved area coverage of high frequencies along with other recognized benefits of corner placement.

4) For reproducing two-track source material, an eclectic system must include the wide speaker spacing afforded by corner placement and also include the center channel.

Appendix

Angular tilt of the speakers has not been discussed. The speakers used exhibited about an 80° polar pattern which was free of sharp beams. Rotating the speakers through plus or minus 20° produced no audible change. If a speaker displays a sharply-defined beam of high frequencies, a temporary expedient would be to rotate the speaker to point the beam toward or away from a listener, but the ultimate solution would be to obtain midrange and tweeter units with "floodlight" rather than "spotlight" radiation pattern, in which case the orientation should be such that the radiation fan covers the room. This usually implies a 45° orientation in a natural room corner.

A Study of a Two-Channel Cylindrical PZT** Ceramic Transducer for Use in Stereo Phonograph Cartridges*

CARMEN P. GERMANO†

***Summary*—An analytical as well as experimental evaluation of the electromechanical equivalent circuit constants of a two-channel flexural-type element is presented. This unit is a hollow cylindrical PZT ceramic structure electroded and polarized to be responsive to two signals perpendicular to each other.**

The electromechanical equivalent circuit chosen to represent this element is based on the analogy between mechanical and electrical vibrating systems, and is a modification of the electromechanical circuit proposed by Mason. It is made up of lumped electrical and mechanical parameters in combination with an ideal transformation ratio.

Along with this evaluation, a brief discussion of performance characteristics of an experimental cartridge utilizing this element will be presented.

Introduction

SINCE the advent of stereo disk recording and reproduction, there has been a steady influx of piezoelectric type cartridges. Largely, these units utilize two active elements to resolve the complex groove motion into the two desired signals. These cartridges have worked well, and design improvements are continually being made. It is the purpose of this paper to discuss a new type of piezoelectric ceramic single-element structure that is responsive to two signals directed at right angles to each other. The design advantages of a single-element stereo cartridge over a two-element unit are obvious and need not be discussed. One manufacturer has introduced such a cartridge and it was reported[1] at the 10th Annual Meeting of the Audio Engineering Society in 1958.

The element in question is a hollow right circular cylindrical PZT ceramic fully electroded on the inside surface. The outside surface has four electrodes running the length of the element but spaced at 90° centers with substantial margins between them. The element is so polarized to produce two parallel-type flexure elements orthogonal to each other.

While the so-called 45–45 system of stereo disk recording has been officially adopted, it should be pointed

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** PZT—Trademark, Clevite Corp.

* Manuscript received by the PGA, March 6, 1959. Revised manuscript received, April 20, 1959. Presented at the IRE National Convention, New York, N. Y., March 24, 1959.

† Clevite Electronic Components, Clevite Corp., Cleveland, Ohio.

[1] J. F. Wood, "A single element stereophonic cartridge," presented at the Tenth Annual Meeting, Audio Engrg. Soc., New York, N. Y.; September 29–October 3, 1958.

out that because of full symmetry, the subject element can be used for both this and the lateral-vertical system. It is only necessary to orient the element properly.

The discussion of the two-channel element is chiefly concerned with an analytical and experimental determination of its performance characteristics as an element. However, it also includes an evaluation of an experimental stereo cartridge utilizing the element.

Electromechanical Equivalent Circuit

Fig. 1 shows the idealized cross-sectional view of the subject element with both the poling and signal connections indicated. The performance characteristics are influenced by the dimensions as well as the ratio of the outer to inner diameter.

It should be pointed out that the analysis to follow is far from rigorous; however, the treatment given does yield results essentially in agreement with those obtained experimentally.

The equivalent circuit chosen for ideally mounted benders is shown in Fig. 2, and is based on the analogy between mechanical and electrical vibrating systems and is a modification of the circuit proposed by Mason[2] in 1935. The circuit is made up of lumped electrical and mechanical elements in combination with an ideal transformation ratio, N. The ideal transducer has the property of converting voltage, current or charge input to force, velocity or displacement output and vice versa. Since the idealized circuit element also transforms mechanical to electrical or electrical to mechanical impedance, a knowledge of the dielectric, elastic and piezoelectric constants of the ceramic is necessary.

Derivation of Circuit Constants

The constants of Fig. 2 are functions of the dimensions of the element and the coefficients of proportionality include the known dielectric, elastic and piezoelectric properties of the material. We define

- C_e, as the free or static capacity of the element.
- N, the transducer ratio, the open circuit voltage as a function of the applied force.
- C_m, the compliance, as the ratio of maximum deflection to applied load.
- m, the effective mass.

Electrical Capacity C_e

For the determination of capacity, we first make use of the capacitance (C) of a cylindrical capacitor which is given by

$$C = \frac{2\pi\epsilon l}{\ln d_0/d_i} \tag{1}$$

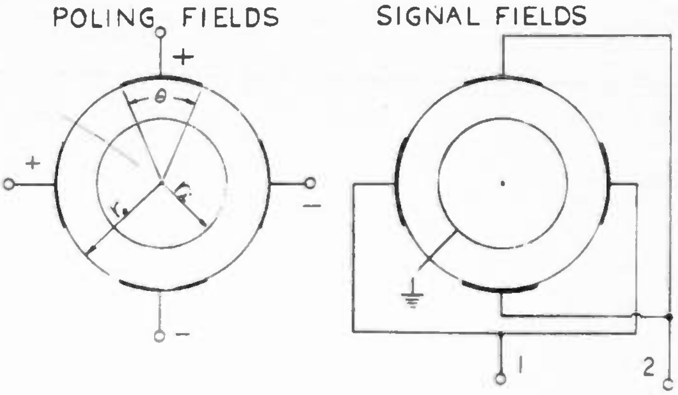

Fig. 1—Idealized cross-sectional view of cylindrical element.

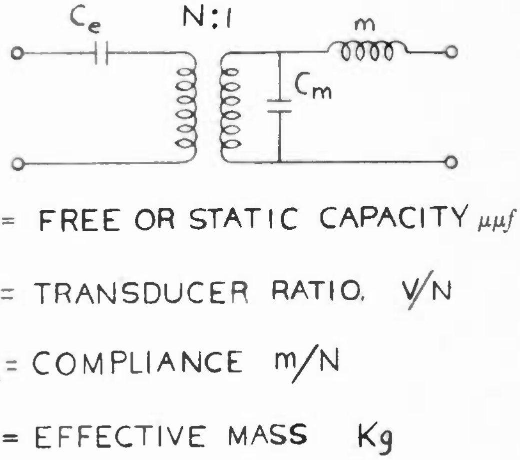

Fig. 2—Electromechanical equivalent circuit.

where

$\epsilon = K\epsilon_0$ = absolute dielectric constant
K = relative dielectric coefficient
ϵ_0 = absolute dielectric constant of free space.

Since we are interested only in one section electroded over an arc subtended by an angle θ, the capacity will be, neglecting fringing, $\theta/2\pi$ times that of the completely electroded tube.

Thus

$$C_e = \frac{K\epsilon_0\theta l}{\ln d_0/d_i}. \tag{2}$$

For two such sections in parallel (using opposite sections)

$$C_e = \frac{2K\epsilon_0\theta l}{\ln d_0/d_i}. \tag{3}$$

Transducer Ratio, N

As defined above, the transducer ratio is a measure of the open circuit voltage V developed per unit of applied force F. Use is made of the theory of flexure to determine the quantity, and thus the following assumptions are made:

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[2] W. P. Mason, "An electromechanical representation of a piezoelectric crystal used as a transducer," Proc. IRE, vol. 23, pp. 1252–1263; October, 1935.

1) The material is homogeneous.
2) The moduli of elasticity in tension and compression are equal.
3) The element is straight.
4) The cross section is uniform.
5) The load is applied perpendicular to the axis of the beam.
6) The span-to-diameter ratio is at least 15 to 1.
7) The maximum stress does not exceed the proportional limit.

Fig. 3 shows a cantilever-mounted cylindrical element along with its bending moment diagram. The flexure formula is given as

$$S = \frac{Mz}{I}. \tag{4}$$

where

S = unit stress on a fiber
M = bending moment
z = distance of that fiber from the neutral axis
I = moment of inertia of cross section with respect to its neutral axis

$$= \frac{\pi}{64}(d_0^4 - d_i^4). \tag{5}$$

The maximum moment of a cantilever beam is Fl, and at any distance x along the length it is Fx.

The charge dQ developed over an area da is given by

$$dQ/da = d_{31}\overline{S}_z, \tag{6}$$

where

d_{31} = piezoelectric modulus relating charge density to stress—field perpendicular to strain,
$\overline{S}_z$ = stress developed averaged with respect to wall thickness;

$$\overline{S}_z = \int_{r_i}^{r_0} \frac{S_z dr}{r_0 - r_i} \tag{7}$$

$$= \int_{r_i}^{r_0} \frac{Fxzdr}{I(r_0 - r_i)}. \tag{8}$$

If we assume that $z = r$, which is valid for small values of θ, we have

$$\overline{S}_z = \int_{r_i}^{r_0} \frac{Fxrdr}{I(r_0 - r_i)}. \tag{9}$$

$$= \frac{Fx}{2I}(r_0 + r_i), \tag{10}$$

$$= \frac{Fx}{4I}(d_0 + d_i), \tag{11}$$

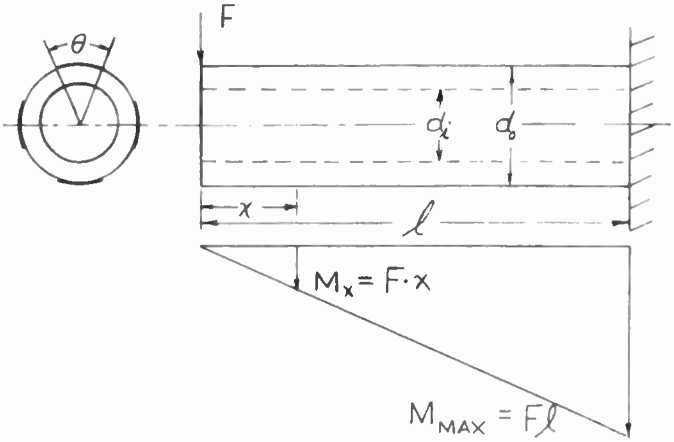

Fig. 3—Cantilever-mounted unit and bending moment.

and

da = the element of area over which the charge is developed

$$= \frac{d_0 + d_i}{4} dxd\theta. \tag{12}$$

Thus

$$Q = \frac{F}{16I}(d_0 + d_i)^2 d_{31} \int_0^l \int_0^\theta xdxd\theta \tag{13}$$

$$= \frac{2}{\pi} F\theta l^2 d_{31} \frac{(d_0 + d_i)^2}{(d_0^4 - d_i^4)}. \tag{14}$$

Since

$$V = Q/C_e \quad \text{and} \quad N = V/F,$$

upon dividing (14) by (2), we have:

$$N = \frac{2}{\pi} \frac{d_{31}}{K\epsilon_0} l \frac{(d_0 + d_i)^2}{(d_0^4 - d_i^4)} \ln d_0/d_i, \tag{15}$$

or

$$N = \frac{2}{\pi} g_{31} l \frac{(d_0 + d_i)^2}{(d_0^4 - d_i^4)} \ln d_0/d_i, \tag{16}$$

where

$g_{31} = \dfrac{d_{31}}{K\epsilon_0}$ = piezoelectric modulus relating open circuit field to applied stress.

Compliance, C_m

Since the compliance is defined as the ratio of maximum deflection to applied force, use is made of the expression for the deflection of a cantilever beam.

$$y = Fl^3/3EI, \tag{17}$$

y = deflection, E = Young's Mod.,

$$C_m = y/F = \frac{64}{3\pi E} \frac{l^3}{(d_0^4 - d_i^4)}. \tag{18}$$

Effective Mass m

Since it is known that

$$\omega_r^2 = \frac{1}{mC_m}, \tag{19}$$

where

$$f_r = \text{flexural resonant frequency}$$

$$= \frac{0.5596}{l^2}\left(\frac{E}{\rho}\,\mathcal{R}^2\right)^{1/2}, \tag{20}$$

$$\mathcal{R} = \text{radius of gyration}$$

$$= (I/A)^{1/2}, \tag{21}$$

$$A = \text{cross-sectional area},$$

$$\mathcal{R} = \frac{(d_0^2 + d_i^2)^{1/2}}{4}, \tag{22}$$

$$\rho = \text{density},$$

$$f_r = \frac{0.14}{l^2}(d_0^2 + d_i^2)^{1/2}(E/\rho)^{1/2}, \tag{23}$$

$$m = \frac{1}{(2\pi f_r)^2}\,C_m, \tag{24}$$

$$= 0.2\rho l(d_0^2 - d_i^2). \tag{25}$$

The complete set of expressions is summarized in Table I for both the cantilever and simple beam mountings. These are more frequently used because simulating ideal mounting conditions in the laboratory is more readily accomplished for simply-supported elements.

EXPERIMENTAL RESULTS

For a given cross section, the results shown in Table I can be reduced to functions of length only. The resulting expressions are shown in Table II for the simply-supported case only. For comparison purposes, the experimentally-determined values are also shown. All of the parameters with the exception of effective mass are easily determined for the simply-supported case. The effective mass is calculated from the compliance and resonant frequency. Appropriate conversion relationships are then applied to obtain the same constants for cantilever-mounted units. For a description of the measurement techniques used, it is suggested that the literature[3] be consulted.

Correlation is seen to be good. The largest discrepancies are found in the expressions for transducer ratio and electrical capacitance. This is not surprising since the assumptions made in the analysis are somewhat idealistic. However, it is interesting to note that the product of these two parameters (or charge sensitivities) are in agreement.

TABLE I

EQUIVALENT CIRCUIT CONSTANTS OF CYLINDRICAL BENDER

Simply Supported	Cantilever Mounted
$N = \frac{1}{2\pi} g_{31} l \frac{(d_0 + d_i)^2}{(d_i^4 - d_i^4)} \ln \frac{d_0}{d_i}$	× 4
$C_e = 2K\epsilon_0\theta l \frac{1}{\ln d_0/d_i}$	Same
$C_m = \frac{4}{3\pi} \frac{l^3}{E(d_0^4 - d_i^4)}$	× 16
$m = 0.4\rho l(d_0^2 - d_i^2)$	× 0.5
$f_r = 0.4 \frac{(d_0^2 + d_i^2)^{1/2}}{l^2} (E/\rho)^{1/2}$	× $(1/8)^{1/2}$

TABLE II

THEORETICAL AND EXPERIMENTAL RESULTS FOR SIMPLY-SUPPORTED CASE

Simply Supported	
$g_{31} = 10.6$ m V-m/N; $K = 1750$; $E = 5.5 \times 10^{10}$ N/m²; e = 7.6×10^3 Kg/m³; $d_0 = 0.05''$; $d_i = 0.03''$; $\theta = 0.6$ RAD	
Theoretical	Experimental
$N = 40l$ V/N	$N = 33l$ V/N
$C_e = 925l\ \mu\mu$f	$C_e = 1130l\ \mu\mu$f
$C_m = 56 \times 10^{-6} l^3$ m/N	$C_m = 55 \times 10^{-6} l^3$ m/N
$m = 80 \times 10^{-6} l$ Kg	$m = 73.5 \times 10^{-6} l$ Kg
$f_r = \frac{2.4}{l^2}$ Kc	$f_r = \frac{2.5}{l^2}$ Kc
l in inches	

SEPARATION

A few words are necessary regarding the matter of separation of channels. Theoretically, there should be no response from the channel orthogonal to the driven one except for a small amount due to interelectrode capacitance coupling. The reason for this will be made clear with the aid of the following explanation.

Consider the ideal situation of perfect symmetry and uniform cross section as shown in Fig. 4.

Channel 1 will develop a signal as shown with a net voltage of say $V = NF$ volts. Channel 2, however, will have equally-opposite charges developed over each section because the portion above the neutral plane is stressed in an opposite direction to that below. The net result is 0 signal.

Practically, however, we do not have complete homogeneity or symmetry and some charge, in addition to that mentioned above, is developed. Typical values of low-frequency separation on a sample lot of elements were from 25 to 40 db. High-frequency values were not determined.

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[3] C. P. Germano, "Study of the Multimorph, a new and improved ceramic element for use in phonograph pickup applications," *Proc. Natl. Electronics Conf.*, vol. 12, pp. 677–688; October, 1956.

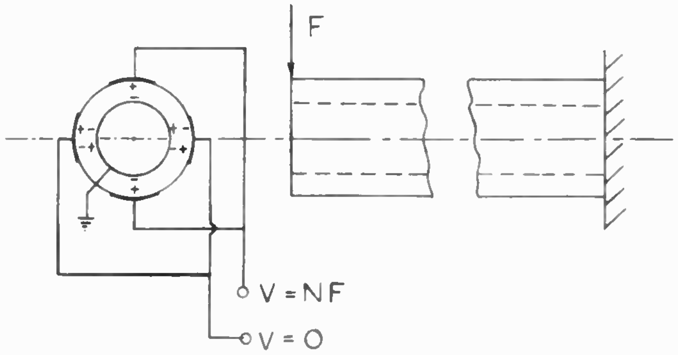

Fig. 4—Theoretical output as a function of single-channel drive.

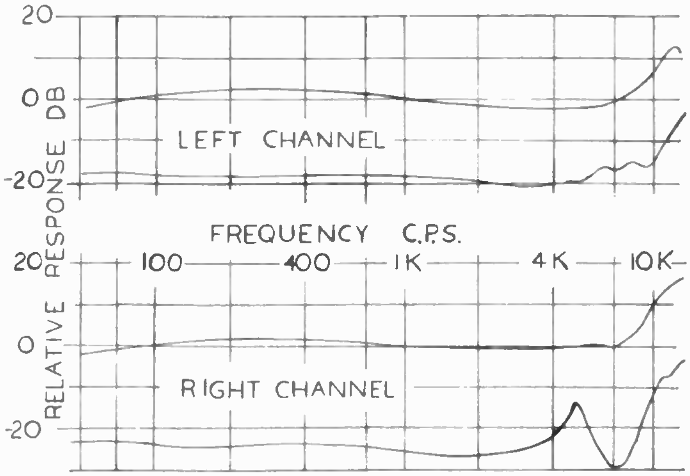

Fig. 5—Response and separation of experimental cartridge.

EXPERIMENTAL CARTRIDGE

An experimental stereo pickup was constructed utilizing the subject element. No real effort was made to completely design the unit around the element; rather, it was found convenient to use parts from a commercially-available monophonic-phonograph cartridge. Some modifications were in order as the cross section of the element in question, not to mention the lead take-off, was totally different than that found in this pickup.

The cylindrical element (of length$=0.5$ inch) was cantilever mounted. A piece of vinyl tubing over its free end supported a stylus mounting. This yielded a stylus point compliance of approximately 2×10^{-6} centimeters per dyne. A pad located midway between the back, or "clamping" pad, and the stylus structure provided some damping. Additional so-called "dead" compliance could have been introduced at the drive end; otherwise, the stylus point compliance could be increased by the use of a thinner-walled or longer cylinder.

The response and crosstalk of this cartridge were determined by the use of the Westrex 1A stereo test record and are shown in Fig. 5. This record is recorded at substantially constant velocity. Thus an amplitude responsive unit without equalization would yield an output decreasing at a 6-db per octave rate. Since the pickup was designed to reproduce the RIAA recording characteristic, the curve is corrected and replotted showing deviation from flat response.

The response from both channels is reasonably flat to 8 kc at which point it begins to rise appreciably reaching a peak beyond 12 kc. This peak is probably a result of the stylus point mass resonating with the record compliance. Some damping could be introduced at the stylus structure to reduce this peak, but this might result in a loss of separation. Because more than an adequate amount of separation is present, some sacrifice could be tolerated. However, it is probable that more effective results could be obtained with other types of stylus mountings.

ACKNOWLEDGMENT

The author is indebted to E. Maciag and E. Johnson for their contributions to this work.

A Frame-Grid Audio Pentode for Stereo Output*

J. L. McKAIN† AND R. E. SCHWAB†

Summary—The development and performance capabilities of a dual pentode using a single cathode, two separate *Framelok* grids and a twin-plate structure contained in one envelope is described. This new pentode, known as Type 6DY7, is a high-performance tube with superior characteristics of uniformity and stability obtained from its unique structure. Such factors as greater uniformity in tube-to-tube characteristics, reduced characteristic spread, and less susceptibility to characteristic deterioration at high dissipations can be obtained.

This dual pentode offers extreme flexibility in application. Three basic configurations are: 1) sections operated separately (single-ended) giving 5 watts of audio power per section; 2) two sections in push-pull, Class AB$_1$ providing up to 20 watts output at less than 3 per cent distortion; and 3) two tubes in push-pull parallel.

A single tube can be used for two stereo output channels, or two tubes can be operated in push-pull for higher power requirements. The same advantages can be used for monophonic audio systems.

The tube, therefore, offers the circuit designer a choice of usage not possible in presently-available tubes and at cost advantages realizable through a reduction in the number of circuit components.

A NEW development, known as the Framelok grid was introduced in March, 1958, and paves the way for a true frame-grid structure which can be utilized competitively in the commercial market.[1,2] We will discuss the development of this grid, its inherent advantages in regard to tube manufacture in general, and finally its application in a dual-pentode audio-output tube for stereo applications.

The frame-grid has been used for many years in a limited number of electron-tube structures where regular grids failed to meet the necessary requirements. Due primarily to its high cost, the use of this type grid has been restricted to such special applications as the ceramic stacked tubes or VHF-UHF types of either planar- or conventional-envelope configuration.

ADVANTAGES OF THE FRAME-GRID

The frame-grid design offers several inherent features not available in a conventional siderod grid. In the frame-grid design, each lateral wire is short and fastened firmly to the frame; thus, the frame supports the lateral wires. The converse is true for conventional grids. This relationship becomes a major factor in making grids with lateral wires of 0.001 inch or less for miniature tubes, and wires of 0.002 inch or less for the larger tubes. Since the frame-grid derives its strength from a solid frame, it is much less limited in regard to lateral wire size, TPI (turns per inch), and close interelement spacings. The increased strength provided by the frame makes it possible to design tubes to withstand severe mechanical shock and vibration.

These inherent advantages of the frame-grid structure, along with its successful use in the stacked-tube designs, emphasized the desirability of applying such a development to commercial tube types. The most logical applications appeared to be the larger beam-power-output and deflection tubes where grid dissipation capabilities, Ib-to-Ic2 ratio, and tube characteristic variations are important factors. Toward this end, a frame-grid development program was initiated which, in 1958, was to result in the current Framelok grid.

DEVELOPMENT OF THE FRAMELOK GRID

Early production techniques for frame-grids involved winding lateral wire around two frames placed back to back. The frames were usually molybdenum, and the laterals were attached by brazing. The frames were then separated, resulting in two half-grids which were mounted on opposite sides of the cathode. A connecting strap then provided the electrical connection between grid sections. From this basic construction, the design progressed toward the ultimate objective of providing a frame-grid, with its inherent advantage, which could be incorporated in commercial tube types.

To get the frame-grid out of the laboratory and into production, several major changes in the conventional frame-grid were necessary. Brazing the lateral wires to the frames was too slow and costly for high production. Also the high cost and poor availability and workability of molybdenum made it necessary to find some other material to be used for frames. The most logical choice of material was nickel-plated steel. Since this material could not be brazed, another method of fastening the lateral wires to the frame was required. At this point it was decided to introduce one other requirement into our design. Not only would we look for a new method of attaching the lateral wire, but for a method which would permit each lateral to fall into planes perpendicular to the side of the frame. This would directly attack a long-standing problem associated with pentode amplifiers, that of alignment between the control and screen grids. In a parallel-planar arrangement, laterals of both grids could be easily aligned across their length. Accordingly, the grid design was changed to facilitate automatic lineup of control and screen grids.

As a result of investigation, the notch-and-peen method was adopted to fasten each individual lateral

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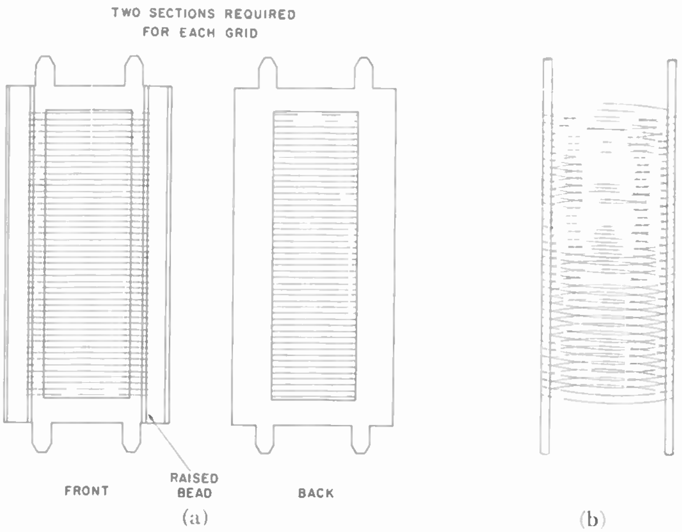
† Sylvania Electronic Products, Inc., Emporium, Pa.

[1] C. Droppa, "Improvements in deflection amplifier design," 1958 IRE NATIONAL CONVENTION RECORD, pt. 7, pp. 147–153.

[2] C. Droppa, "Improving the deflection amplifier," *Electronic Ind.*, vol. 17, pp. 76–79; May, 1958.

wire straight across the frame. A bump or bead projecting above the frame surface was incorporated to receive the wire. The size and position of the bead is governed by the wire size, the TPI of the grid, and the thickness of the frame. Fig. 1 shows these construction features and provides a comparison between the Framelok grid and a conventional siderod grid. Notice the greater mass of the frame. It provides a much larger heat sink than the siderods of a conventional grid. Also note the bump or bead into which the lateral wire is anchored under tension by the notch-and-peen operation. This tension, plus the heavy frame, eliminates wire sag and grid bowing and thus eliminates two of the common causes of shorts with conventional grids.

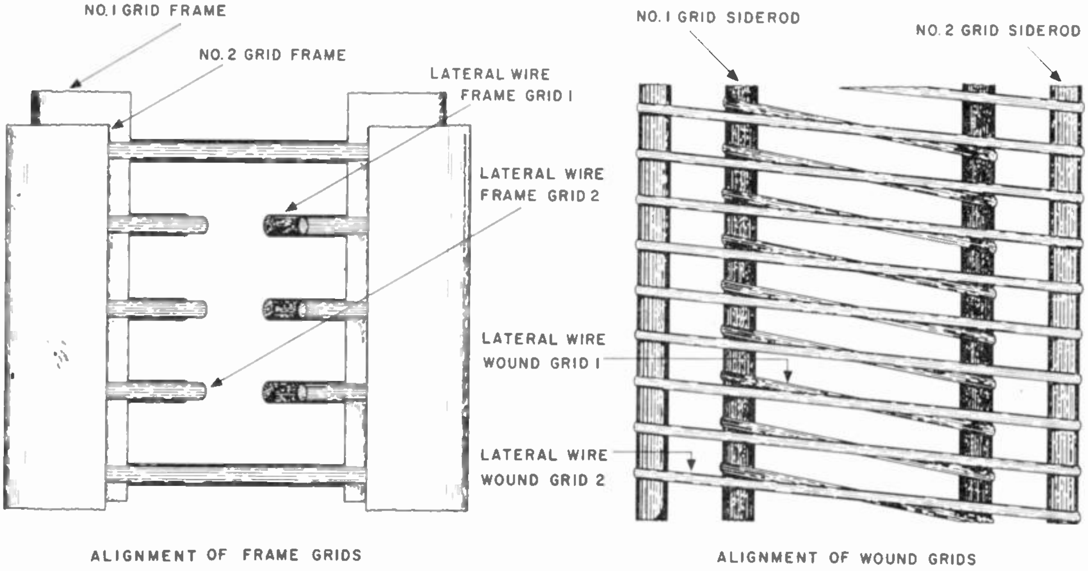


(a) (b)

Fig. 1—Comparison of screen grids. (a) Frame-grid. (b) Conventional grid.

Inherent Advantages of the Framelok Design

Fig. 2 shows how the Framelok design has eliminated the pitch of wound grids. Fig. 2(b) illustrates that, even with equal turns per inch of control grid and screen grid, conventional wound grids will not be perfectly aligned except at the center. This is due to the fact that the major diameters of these grids are different, and this results in the grid laterals having different slopes. It should be pointed out that the grid cross-over has been exaggerated for illustration. Note in Fig. 2(a) that the grid lineup problem has been eliminated. By making the wire sizes slightly different, the screen grid can be made to be entirely in the shadow of the control grid, and ideal alignment exists over the entire length of each lateral wire. This type of alignment will provide a much more favorable plate-to-screen-current ratio that is available from conventional grids.



(a) (b)

Fig. 2—Comparison of conventional and frame-grid alignment. (a) Alignment of frame-grids. (b) Alignment of wound grids.

It may appear that the perfect alignment previously mentioned may be beyond control in production. With examination of Fig. 3, it becomes apparent that this

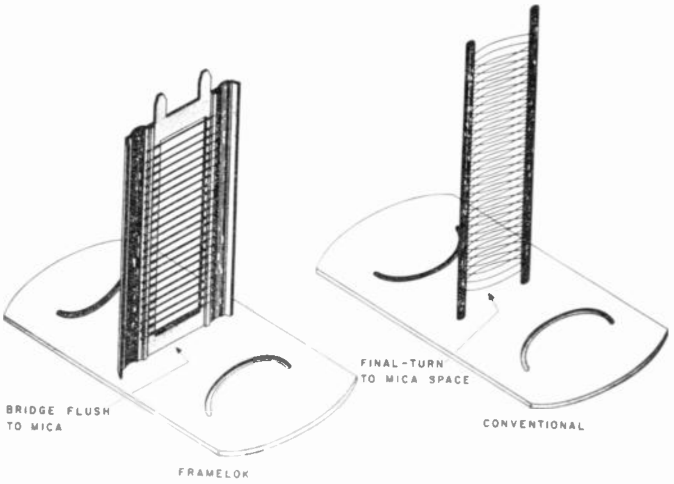


Fig. 3—Grid to mica relation.

factor is actually controlled with the grid manufacture. Because the bridge of the frame rests against the mica, each grid is automatically positioned during assembly.

The fact that the bridge is positioned against the mica (Fig. 3) is also a major factor in cutoff control. This bridge can be regarded as an extension of the grid. Since this is the area most susceptible to uncontrolled conduction with conventional grids, the bridge becomes important electrically as well as mechanically. Using a conventional grid, the electron flow near the mica is controlled primarily by the last turn. Any deviation from the desired location or any distortion of this wire will have an adverse effect on cutoff. With the solid bridge of the frame-grid performing the function of the last turn of a conventional grid, much of this variation is eliminated and cutoff control is improved.

The greater rigidity of the frame-grid is also an important factor in the control of grid spacings. The heavy frame permits a much tighter grid-to-mica fit with less grid distortion during assembly than can be obtained with siderod grids. As a result, the spacings between elements are more uniform than in conventional tubes. The technique of manufacturing the frame-grid in two parts is also an important factor in more uniform spacings. Each half of the frame-grid is mounted separately, and thus the spacings on one side of the cathode are independent of the spacings on the other side. With a conventional grid, factors which cause a change in the spacings on one side of the cathode often cause an opposite change on the other side. These advantages of the frame-grid are of primary importance since the control of grid-cathode spacings is a major factor in tube characteristics uniformity.

At this point, we had developed a frame-grid with the desired inherent mechanical advantages. By virtue of its unique design and mechanical superiority, the Framelok grid could be utilized to improve grid lineup, provide a closer control of grid spacings, and thereby improve the uniformity of tube characteristics. To make this grid practical for commercial tube types, a machine was designed to manufacture it economically in production quantities.

Application of the Framelok Design

With the grid design and production machinery developed, the next question was which type of application could best utilize the advantages of the Framelok grid. The most immediate need appeared to be in the field of TV horizontal deflection. The Type 6FH6 was thus introduced. The fast onset of Hi-Fi and Stereo has also emphasized the need for a dual-pentode power-output tube. As a result, the Type 6DY7 is the second Sylvania development using the Framelok principle.

Fig. 4 is a cross section of the Type 6FH6 tube. A study of this structure will show the possibility of converting this single pentode to a dual pentode, even with the limited number of pins available in the octal base.

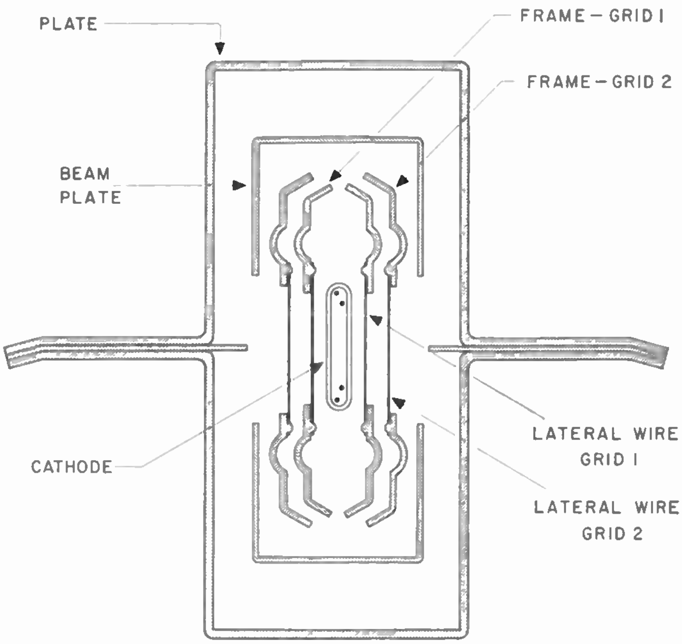


Fig. 4—Cross section of mount employing frame-grids.

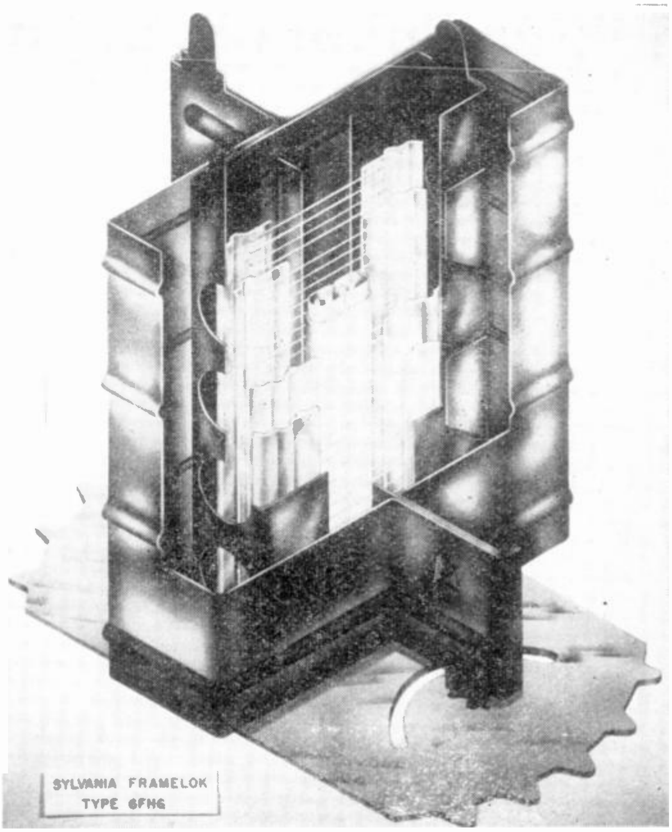


Fig. 5—Framelok structure of Sylvania 6FH6.

If the plate structure is split to provide two identical plates and the control-grid connecting strap eliminated, the result is a dual pentode with a single cathode, two separate Framelok control grids and a common screen grid. This is the basic configuration of the Sylvania Type 6DY7 in a T-12 bulb. Fig. 5, again using the Type

6FH6 as a typical Framelok structure, shows a cutaway view of a Framelok mount and the physical relationships of the tube elements are apparent.

PERFORMANCE OF THE 6DY7

We now have a practical dual-pentode structure using the Sylvania Framelok grid. The Type 6DY7 is designed to provide up to 20 watts of output at less than 3 per cent distortion under Class AB_1 push-pull operation. Taking first things first, Fig. 6 shows the average characteristics and typical performance data for single-section Class A_1 operation. These data show a power output of 5 watts at 9.0 per cent total harmonic distortion. Thus, one Type 6DY7 tube can be used in a stereo amplifier to provide 5 watts of audio output per channel. It should be pointed out here that, consistent with standard tube rating practice, these power output and distortion figures were obtained with a shunt-fed resistive load and are, therefore, representative of the power available at the plate of the tube. They do not reflect any losses which may be encountered in a typical output transformer. To provide direct comparisons with existing published data, all power output and distortion measurements shown will be consistent with this practice.

Fig. 7 shows data for the Type 6DY7 in a Class AB_1 push-pull amplifier under two typical operating conditions. With the plate voltage limited to 250 v, and a plate-to-plate load resistance of 9000 ohms, a power output of 11 watts with 2.5 per cent total harmonic distortion is available. If the maximum allowable plate voltage of 400 v is used with a plate-to-plate load resistance of 14,000 ohms, the maximum signal power output is 20 watts with only 2.0 per cent distortion.

The data presented in Figs. 6 and 7 have shown the typical characteristics of the Type 6DY7, but how does this new development compare with existing types? For comparison purposes, Fig. 8 shows the characteristics of Type 6DY7 along with Types 6V6 and 6BQ5 under similar voltage conditions. Note that at the Eb = Ec2 = 250-v conditions, the Type 6DY7, compared with the Type 6V6, provides 10 per cent more power output, greater plate efficiency, and a lower total harmonic distortion. In comparison with the 6BQ5, the power output is comparable with improved distortion.

Since the Type 6DY7 is similar for power output and distortion to the Types 6V6 and 6BQ5, a comparison in costs becomes a primary consideration. It is apparent that this concept of incorporating two identical pentodes in one envelope makes possible the use of a single Framelok Type 6DY7 in place of two conventional audio output tubes and still deliver the same output. With one socket serving as two, the equipment manufacturers can reduce costs by reducing production wiring time, eliminating components and hardware, and substantially cutting tube inventory. These savings should provide a lower net cost as well as a possible reduction in package size and weight.

Plate voltage	250 volts
Grid no. 2 voltage	250 volts
Grid no. 1 voltage	−12.5 volts
Peak AF signal voltage	12.5 volts
Zero signal plate current	50 ma
Maximum signal plate current	45 ma
Zero signal grid no. 2 current	3.0 ma
Maximum signal grid no. 2 current	9.0 ma
Transconductance	6,000 μmhos
Plate resistance (approx.)	28,000 ohms
Load resistance	5,000 ohms
Maximum signal power output	5.0 watts
Total harmonic distortion	9.0 per cent

Fig. 6—Class A_1 characteristics (single section).

Plate voltage	250	400 volts
Grid no. 2 voltage	250	250 volts
Grid no. 1 voltage	−16	−20 volts
Peak AF grid-to-grid voltage	32	40 volts
Zero signal plate current	77	58 ma
Maximum signal plate current	74	74 ma
Zero signal grid no. 2 current	3.5	1.7 ma
Maximum signal grid no. 2 current	15.5	14.0 ma
Load resistance (plate-to-plate)	9,000	14,000 ohms
Maximum signal power output	11	20 watts
Total harmonic distortion	2.5	2.0 per cent

Fig. 7—Class AB_1 characteristics (two sections, push-pull).

	6DY7	*6V6GT*	*6BQ5*
Plate voltage	250	250	250
Grid no. 2 voltage	250	250	250
Grid no. 1 voltage	−16	−15	—
Rk	—	—	130 ohms
Peak AF grid-to-grid voltage	32	30	22.6
Zero-signal plate current	77	70	62 ma
Maximum-signal plate current	74	79	75 ma
Zero-signal grid no. 2 current	3.5	5	7 ma
Maximum signal grid no. 2 current	15.5	13	15 ma
Load resistance (plate-to-plate)	9,000	10,000	8,000 ohms
Maximum signal power output	11	10	11 watts
Total harmonic distortion	2.5	5	3 per cent

Fig. 8—Class AB_1 amplifier (one 6DY7 or two 6V6 or 6BQ5 in push-pull).

FLEXIBILITY IN APPLICATION OF THE 6DY7

In addition to being a high-performance tube with superior characteristics of uniformity and stability, the Type 6DY7 offers extreme flexibility in application. This is particularly true in the stereophonic field. Although this dual pentode was designed for application in stereophonic sound systems, it is just as useful in monophonic applications. The benefits and features this new tube makes available are applicable to all audio equipment.

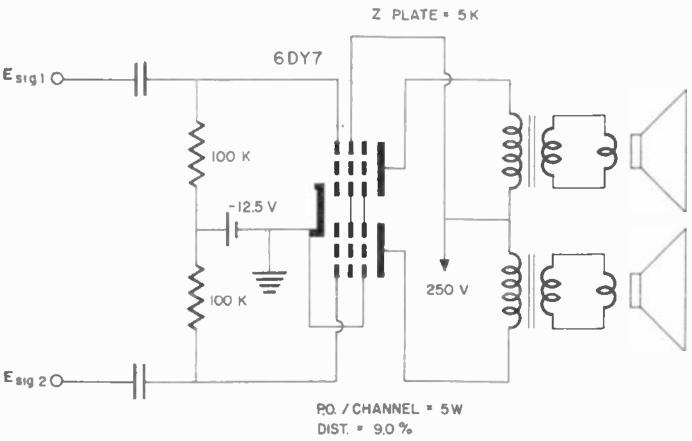
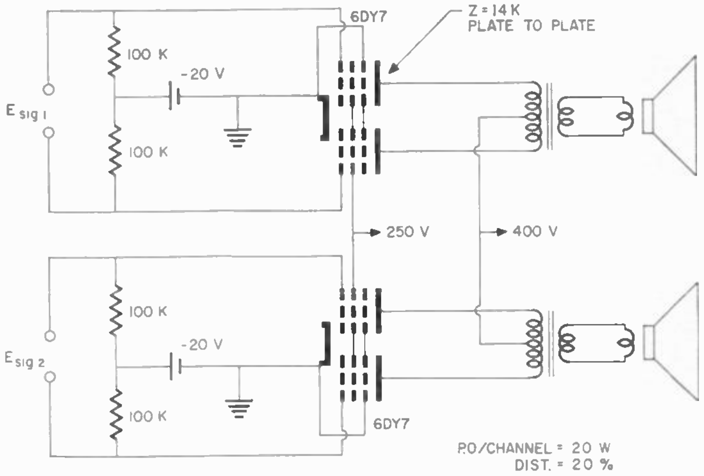
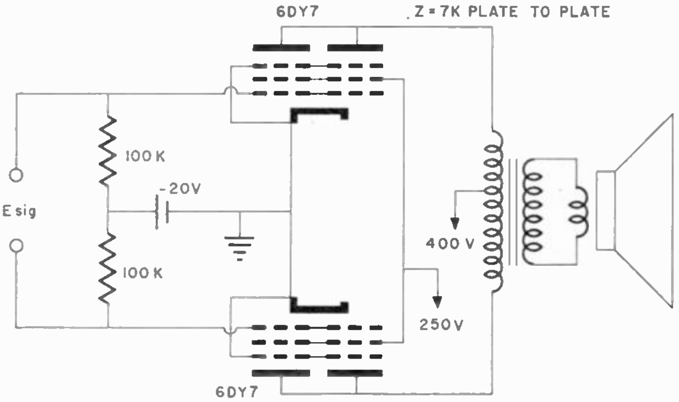


Fig. 9—Two stereo channels—one tube.



	W_p	W_{c2}
ZERO SIG	116	0.2
MAX SIG	148	1.7

Fig. 10—Two stereo push-pull outputs—two tubes.



	W_p	W_{c2}
ZERO SIG	37.0	0.75
MAX. SIG	18.0	7.0

P.O. = 40 W
DIST. = 3 %

Fig. 11—Push-pull parallel outputs—two tubes.

Fig. 9 shows an example of the flexibility of this dual pentode. Normally two tubes—one for each channel—are required to provide the audio power for a basic low-output stereo system. In the case of the Type 6DY7, each pentode operates single-ended, permitting a single tube to perform the job. In this type of application, approximately 5 watts of audio power per section is available at less than 10 per cent total harmonic distortion. In consideration of a dual tube for an application such as this, the first question likely to be asked is what degree of cross modulation will be encountered. Laboratory measurements have been made with each pentode carrying a separate signal or channel. At this time, no difficulty has been encountered from cross modulation when fixed bias is employed. Data obtained to date indicate that with each section operating at maximum power output, unwanted signals above 300 cps at the plate of the section in question are 50 db below the desired signal. A greater degree of cross modulation has been encountered at frequencies below approximately 300 cps when self bias is employed. The increased effect of cross modulation at frequencies below 300 cps should not be a critical factor in stereo applications, since direction is difficult to determine at these frequencies.

With the modern trend toward high-power amplifiers for the Hi-Fi market, it is reasonable to assume stereo will follow the same path. Fig. 10 shows how the Type 6DY7 will lend itself to this type of application. By operating the two sections in push-pull, a single tube will provide up to 20 watts of audio power at less than 3 per cent total harmonic distortion. Thus, two tubes, one for each channel, will provide the same service obtained from four tubes of conventional design. In regard to push-pull operation, the incorporation of two pentodes in one envelope will undoubtedly raise the question of balance between sections. It is obvious that with this type of construction, matching pairs of tubes for push-pull will not be possible. However, with the exception of the case where tubes are specially selected, the increased uniformity provided by the Framelok design will result in closer balance than a random selection of two conventional tubes. At this time it appears that, in all but the most critical applications, plate current unbalance will not be a problem.

A third configuration which can be fulfilled by this Framelok audio output tube is shown in Fig. 11. The Type 6DY7 allows application of two tubes in push-pull parallel where four tubes of conventional design are required. This type of service, which provides up to 40 watts of audio power, will find its primary usage in high-power monophonic systems, but it serves to illustrate the flexibility provided the equipment designer by this development.

Conclusion

It has been shown that the Framelok structure provides a grid whose strength comes from its frame. The grid lateral wires are completely supported by the frame and are not required to contribute to the rigidity of the grid. Since its greater mass dissipates heat more readily, this grid has virtually no warping or bowing. The Framelok grid structure is such that interelement spacings can be more closely controlled, being dependent only on the position of the mica slots. Precise grid alignment is more easily achieved and grid-to-grid duplication is much more exact. These factors all lend themselves to reducing the initial dispersion of electrical characteristics as well as contributing to a lower failure rate. Grids and mica slots are designed for maximum contact area, permitting greater rigidity and providing a more rugged mount. The unusually tight fit of the Framelok grid and mica substantially reduces the possibility of grid movement; this in turn, reduces microphonism and noise.

Utilization of the Framelok grid in a dual-pentode design resulted in the development of the Type 6DY7, which is shown to be a uniform, high-performance audio-output tube with extreme flexibility. This new development has the capability of handling two separate signals with the two identical pentodes, making it exceptionally suitable for stereo applications. Using the two pentodes in Class AB_1 push-pull, the Type 6DY7 is designed to provide up to 20 watts of output at less than three per cent distortion. The many advantages of the Framelok grid structure could be utilized in other power output and deflection tubes where grid dissipation capabilities and screen current characteristics limit performance and serviceability of electron tubes.

Contributors

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Carmen P. Germano (S'49–A'53–M'56–SM'56) was born in Cleveland, Ohio on August 5, 1924. His education at Northwestern University, Evanston, Ill., was interrupted by service in the U. S. Air Force during World War II when he served as a radio operator and repairman. He received the B.S. degree in electrical engineering in 1950.

He has been with Clevite or a predecessor company since April, 1950, having started with the Brush Development Company at that time. As a result of graduate work in the Evening Division of John Carroll University, Cleveland, he received the M.S. degree in physics in 1952. His early work was concerned chiefly with measurements and methods of measurements of piezoelectric elements.

He is presently head of the Electroacoustics Section of Clevite Electronic Components where he is concerned with the design and development of piezoelectric devices in the Audio and Ultrasonic fields. He is also involved in the evaluation of new or improved piezoelectric ceramic and crystal elements.

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James L. McKain (M'53–SM'57) was born in Washington, Kans. on June 28, 1927. Following a period of service with the U. S. Navy beginning in 1945, he entered the University of Nebraska, Lincoln, from which he received the B.S. degree in electrical engineering in 1951. In July of that year he joined the Radio Tube Division of Sylvania Electric Products, Inc., as a student engineer in the General Engineering Department.

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Ray E. Schwab was born at Shippensville, Pa., on July 14, 1923. He entered Purdue University, Lafayette, Ind., in 1941, leaving in 1943 to enter the armed services where he served until 1945, including one year in the European Theatre. In 1946, he returned to Purdue and received the B.S. degree in electrical engineering in February, 1949.

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