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CORRESPONDENCE

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The Editor's Corner

EARLY EXPERIENCES WITH STEREOTAPE RECORDING

ALTHOUGH "do-it-yourself" stereo is now as common as tape recording, we can remember a time when it was almost unheard of. Nearly twenty years ago we became intrigued with the possibility of binaural and stereophonic sound on multi-channel tape. In one of our early experiments we set up a pair of close-spaced microphones and allowed our two-channel recorder to pick up a conglomeration of sounds in the laboratory, including a conversation, a typewriter in the background, some hammering, drilling, etc.

We played back binaurally through earphones, each one on a separate channel; and it was remarkable that a listener could direct his attention to the conversation without being bothered by the background sounds. If we switched to either channel alone, the interfering noises rendered the conversation practically unintelligible. Many applications were considered such as music, conference recording, court reporting, and recording noises in the field for later study in the laboratory. Of all these, the most fascinating was music.

To anyone who heard even the early low fidelity attempts, the sound of binaural music was a miracle. Russ Tinkham, Bob Vaile, Hal Leedy, John Boyers, Ray Zenner, Jack Kemp, Carl Titus, Dave Wiegand, used to spread the gospel with slogans such as: "Binaural sound with a response to 3000 cycles is far more enjoyable than the best 15000 cycle monaural sound." "You don't even notice distortion, wow, or noise that you couldn't tolerate in a single-channel record." "Harvey Fletcher of Bell Labs has predicted that the day will come when all reproduced sound will be stereophonic."

To hasten this day, we built a three channel tape machine that would meet the highest standards of fidelity. (The capstan was the shaft of a half horsepower three phase motor which we rewound for eighteen poles to slow it down.) We then undertook to make a series of live recordings, so that we could study microphone and loudspeaker placement, and room acoustics.

On a Friday night, shortly after we had put our deluxe system together, J. P. Lekas arranged for us to set it up at The Arthur Murray Studios, where they had a dance combo for their social evening. Most of the customers forgot they had come to dance, and watched in amazement as we moved our microphones about, adjusted the patterns of our 639B's and set the levels on our three VU meters. We had to explain many times over what "Stereophonic" meant.

After a few playbacks everyone got the idea. In fact the musicians did some experimenting of their own. The clarinet player would walk from one mike to another blasting out some choice passages as he went by. Everyone predicted that it would sound terrible, but the results were quite pleasing. (We are reminded of this whenever someone complains about the artificial setups and tricks used in making stereo recordings today.) Some very nice demonstration recordings came out of this session.

With the first stereo tape recorder we felt like pioneers in unexplored territory. We were anxious to discover techniques that should be used with microphones, loudspeakers, and room acoustics to obtain best results. Live music was needed, of course; fortunately the Chicago Musical College was close by, and allowed us to record their rehearsals. They arranged what amounted to a music festival with orchestral and choral works, in addition to solo instrument, and vocal renditions. We had several field days, experimenting with all kinds of microphone distances and directional patterns. After each day's recording sessions we would audition the results and outline new experiments for the next day.

When we wanted to record professional artists we had to get permission from Mr. Petrillo and the musicians union. Surprisingly enough, they were cooperative when we explained that it was only for experimental purposes. At the NBC studios in Hollywood we set up our equipment on a hand-truck, and wheeled it from studio to studio, picking up rehearsals such as the Lucky Strike Hit Parade, and the NBC orchestra. The musicians were impressed with binaural playback through earphones. One conductor said that, impossible as it might seem, the reproduced sound was even more pleasant than direct listening. Although we had permission from the union, they made sure that the recordings would remain experimental by stationing a couple of "jokers" near the microphones who made audible remarks at intervals. The remarks were mostly unprintable, but added spice to our playbacks.

Early in 1948 the Chicago Musical College, Armour Research Foundation, and Illinois Institute of Technology ran a "debut," where stereotape was presented to society and to the public. Dr. Rudolph Ganz and other distinguished members of the faculty and of the press sipped tea and listened to stereophonic music.

We got some very favorable writeups in the press, but since no equipment was available on the market, there was not much that the average hi-fi listener could do. At the time we suggested to the recording companies that they ought to preserve the artistry of people like Toscanini on multi-channel stereo tape for future possibilities; but as far as we know, nothing was done until many years later.

Our many and varied experiments gave us a feeling for what could be expected from stereophonic sound. One thing we learned very early was that stereophonic sound was *not* the equivalent of binaural. With binaural you could set up a pair of microphones eight inches apart and get a sensation of realism; no matter whether the microphones were in the orchestra or in the gallery; and in spite of poor room acoustics or poor microphone directionality. You could break every rule and still come out all right.

Not so with stereo. According to the literature, and according to common sense, you were supposed to set the microphones about ten to thirty feet apart depending on the size of orchestra, with an extra microphone in the center if you had a third channel. You were not supposed to manipulate the placement of instruments or soloists, since everything would come out all right when you played back with loudspeakers in the same location as the microphones. This was all standard procedure. Stereo recordings were made this way ever since the word was coined, and are still being made this way today.

Now the first times you played back stereo, especially on location where everyone was flushed with contagious enthusiasm, the recording sounded wonderful. A few days later it didn't seem quite as good. Then you set it up for the panel of critics, and by gosh, it was really terrible. Someone would always say, "the violins are on the right, aren't they." Some one else would disagree. So you would play selected passages, and you would try adjusting the volume levels, changing the speaker polarity, interchanging the left and right channels, or setting tone controls.

As a matter of fact, the violin sounds were coming out of more than one loudspeaker and their location depended on where you were standing. But even if you stood in the middle, their position was indefinite. Whenever the tone-coloring, pitch, or volume of the instruments changed, their location changed also. Sometimes whole sections of the orchestra seemed to fly back and forth across the room, for no reason at all, which was very disconcerting.

It was true that you got a nice blending if the whole orchestra was playing. At this point the suggestion would come up that we should try connecting the same channel to all of the speakers. An A-B test would be arranged, and while most people could notice a difference, few were willing to say which was stereo and which was not. Those who did say were wrong about half the time. We would get into a discussion as to whether there should be a difference anyway, as long as everything was supposed to be blended.

Then there were the two channel versus three channel arguments. We found that an independent third channel was effective for soloists or announcements, especially when played back in a large room, but seemed only to reduce the spread and add to the confusion when used for ensemble work. A center channel obtained by mixing the outer two channels had the same disadvantages, without the advantages, since the microphones were too far apart for intimate pickup of a soloist. Although all our music had been recorded on three separate channels, when we demonstrated it we found that we had to move the center speaker back, and to tone it down both in volume and in high-frequency response to keep it from dominating. Often if could be turned off entirely without much difference.

Our experience convinced us that the usual "common sense" true-stereo method of pickup and playback is ineffective; particularly with all-directional microphones spaced in front of an ensemble, and playback through speakers about the same distance apart as the microphones. There is good reason why it should not work, when we consider how artificial a situation we have created. In normal hearing a person locates a sound source by means of differences in what he hears with each ear: differences in loudness, in phase, and in frequency spectrum. When all these (and other more subtle) clues check and double-check each other, then the brain tells us that a sound is "there."

Stereo loudspeakers can only give us one of these clues: namely loudness. The phase and other clues are wrong, so instead of helping they actually detract and confuse. To compound the confusion, the left-hand loudspeaker is heard not only by the left ear, but also by the right ear about a millisecond later; and similarly for the right-hand speaker. Under these artificial conditions we try to make a best guess as to location, but we have the feeling that something is not right.

Since loudness is our only clue, is this at least always presented correctly to the listener? Unfortunately it is not. The interaction of room acoustics, performer locations, and spaced microphones gives a standing wave pattern which can make the wrong side louder at certain frequencies, causing annoying movements of sound sources which were stationary. Also in the listening environment, when a listener moves appreciably from a central location, the closer speaker will take over because of increased loudness and precedence effect.

To counteract these difficulties the recording engineer uses drastic methods and exaggerates the differences between the channels. Often the methods are so disillusioning that it is a good thing that they are not known by the ultimate user. The purist is invariably horrified; he insists that we don't have stereo here, but an evil concoction by extremists who stop at nothing to give weird, unnatural, over-dramatized effects. But if you let him audition the results, sight unseen, he will most likely choose the one that was made under "questionable" conditions. There is a great deal of room here for recording art. Occasionally a great stereo recording is made, but more often the results are not impressive.

We might summarize our experiences by saying that we found there was no such thing as stereo. But to say that there is no stereo is like saying that there is no Santa Claus to a group of believers, who have faith that somehow, somewhere the good St. Nicholas exists, even though a few of the details sound fishy. Yet this analogy is good. The basic concept of loudspeaker stereo assumes a naive type of listener who hears only what we want him to hear and uses the information in a special way. From slight differences in loudness of separated sources he is supposed to reconstruct an authentic sound image.

Sooner or later we realize that we have something which does not work as we thought it should. But although it is not real, stereo can give beneficial results, especially when modified with due respect to its limitations.

And so it is with Santa Claus.

MARVIN CAMRAS, *Editor*

PGA News

CHAPTER NEWS

Bill Ihde, Chairman of the Chapters Committee, reports that he received a notice that the Michigan Chapter of the Acoustical Socitey of America had a meeting in which the local IRE Chapter cooperated. The speakers were Klaus Walther, Bendix Research Laboratory, who spoke on "Pulse Compression in An Acoustic Wave Guide," and John Wescott of the Acoustics and Seismic Laboratory, University of Michigan, who spoke on "Sources and Characteristics of Sounds at High Altitudes."

Also, notice was forwarded of the Syracuse Chapter being placed on the inactive list. Other Chapters who are considering inactivity, might try what the Michigan and Chicago Chapter have done successfully which is to meet jointly with other PG Groups or Professional Societies when a subject of mutual interest is available. Not only does this cut down on the duplication of meetings, but provides a broader base of members for attendance at the meetings.

Boston

The Boston Chapter of PGA met on December 12, and heard a paper by Alan D. Grinnell of the Harvard Biological Laboratory entitled "Neural Correlates of Echo Location in Bats."

Chicago

On January 16, the Chicago Chapter of PGA had a joint meeting with the Chicago Acoustical and Audio Group. The speaker, George W. Kamperman, of Bolt Beranek and Newman gave a paper entitled "Tranducers for the Measurement of Sound, Vibration and Strain."

Philadelphia

On December 8, the Philadelphia Section had a meeting and heard a paper by Mrs. E. L. R. Corliss and T. Priestly of the National Bureau of Standards, entitled "Studies of Transient Distortion in Hearing Aids/A New Electrostatic Loudspeaker." The Meeting also included a demonstration of loudspeaker response to gated sine waves, and a comparison among several types of loudspeakers.

San Francisco

On December 6, the San Francisco Chapter had a meeting with a speaker, Dr. Vincent Salmon, of the Stanford Research Laboratories. His paper was entitled "Measurement in Architectural Acoustics."

ANNOUNCEMENTS

International Congress on Acoustics

The Fourth International Congress on Acoustics will be held in Copenhagen, Denmark, August 21 to 28, 1962. Further information and registration forms may be obtained by writing to Fourth International Congress on Acoustics, Østervoldgade 10, Copenhagen K, Denmark.

Magnetic Recording Bibliography Available

The Kinelogic Corporation, 1256 North Fair Oaks Ave., Pasadena, Calif. has compiled a "Bibliography of Magnetic Tape Recording, 1954–1960 Inclusive," which may be obtained by writing to the above address. James T. Blakistone, the President, indicates that in the next few months revisions will be made to include a bibliography for 1961, an author index for 1954–1961, plus corrections and supplemental information.

Conference on Recording to be Held in Budapest, Hungary

The Hungarian Society for Optics, Acoustics and Cine-technics is organizing a Conference between October 15–18, 1962, on the subject of "Signal Recording on Moving Magnetic Media."

The papers will treat the theory of magnetic recording and its possible application in

sound recording
video recording
automation
electronic computers and measurement techniques.

At the same time an exhibition will be held at which the various factories and firms will exhibit equipment for the above purposes.

For further information about the Conference, please write to: Optikai, Akusztikai es Filmtechnikai Egyesulet, Szabadsagter 17, Budapest V. Hungary.

A preliminary list of papers, which have been entered conditionally is as follows: (the abstracts of these papers, and 10 more whose titles have not yet been received, are now being obtained.)

It should be noted that this listing is tentative, and that several of the authors listed below have withdrawn their papers.

Eric D. Daniel, Memorex Corp., Santa Clara, Calif., "Printing and Instability Problems in Magnetic Tape."

Takashi Kumakura, Sony Corp., Tokyo, Japan, "Demagnetization in Unbiased Magnetic Tape Recording."

J. Greiner, Deutsche Akedemie der Wissenschaften zu Berlin, DDR, Ger., "Remanenz—Kurven—Theorie oder teilweise ideale Magnetisierung beim Aufzeichnungsvorgang."

Donald F. Eldridge, Memorex Corp., Santa Clara, Calif., "A Special Application of Information Theory to Magnetic Recording."

R. Straubel, Deutsche Akademie der Wissenschaften zu Berlin, DDR, Ger., "Die Erweiterung des Frequenzbandes bei der Abtastung von Magnetogrammen."

K. Willaschek, Deutsche Akademie der Wissenschaften zu Berlin, DDR, Ger., "Über die Abtastung von magnetogrammen mit Spaltsonden."

Z. Vajda, Hungarian Radio and Television, Budapest, "Intermodulation by Short Wavelengths."

Peter C. Goldmark, CBS Laboratories, Stamford, Conn., "High Density Magnetic Sound Recording System."

P. Orgovanyl, "Hunnia" Filmstudio, Budapest, "Transient Verzerrungen bei der magnetischen Schallaufzeichnung."

K. E. Gondesen. Inst. für Rundfunktechnik, GmbH, München, DBR, Ger., "Bildsynchrone Tonaufzeichnung."

Dr. K. A. Mittelstrass, VEB Filmfabrik AGFA Wolfen, DDR, Ger., "Die Herstellung der OIRI—Bezugstander zur Einstellung von Magnetonbandanlagen."

G. Heckenast, Ungarischer Rundfunk und Fernsehfunk, Budapest, "Massapparatur für raashe Kopf- und Bandparametermessungen."

F. Takacs, Ung. Schallplattenfabrik, Budapest, "Untersuchung der Betriebseigenschaften von Magnetbander."

A. Tolk, Betriebslaboratorium für Rundfunk und Fernsehen, Berlin, DDR, Ger., "Elektronische Regelung der Geschwindigkeit von Magnettonlaufwerken."

Göhler, Betriebslaboratorium für Rundfunk und Fernsehen, Berlin, DDR, Ger., "Einige konstruktive Details moderner Magnettonlaufwerke."

V. Stuzzi, Stuzzi Radiotechnische Fabrik, Wien, Österreich, "Entwicklung und Produktion moderner Heimtonbandgerate."

W. Bogen, Wolfgang Bogen GmbH, Berlin-Zehlendorf, DBR, Ger., "Konstruktions- und Fertigungstechnische Probleme von Magnetköpfe."

G. Heckenast—Z. Vajda, Ungarischer Rundfunk und Fernsehfunk, Budapest, "Volltransistorisiertes Studio-Magnettongerat."

H. Völz, Deutsche Akademie der Wissenschaften zu Berlin, DDR, Ger., "Über neueste Arbeiten auf dem Gebiet der Messwertespeicherung."

M. A. Curry, AGFA Incorporated, New York, N. Y., "Die magnetische Speicherung seismischer Signale bei der Lagerstattenforschung."

Gy. Koch, Institut für Geophysik, Budapest, "Gerauscherscheinungen mechanischen Ursprunge der magnetischen Signal-Spreicherung."

G. Badonyi, Institut für Geophysik, Budapest, "Magnetband-Speicherung für geophysikalische Forschungen."

A. S. Hoagland, IBM Corp., San Jose, Calif., "High Density Digital Magnetic Recording."

M. A. Curry, AGFA Incorporated, New York, N. Y., "Die magnetische Speicherungsverfahren der elektronischen Mechenmaschinen."

Dr. D. B. G. Edwards, Electrical Engineering Laboratories, University of Manchester, England, "Design and Use of the Magnetic Tape System on the "Atlas" Computer."

V. Chlouba, Praha, OSR, "Die magnetischen dinamischen Speicher der automatischen Rechenanlage EPOS 1."

W. Neumann, VEB Messgeratewerk Zwönitz, Entwicklungstelle, Berlin, DDR, Ger., "Entwicklung von Magnetbandgeraten zur analogen und digitalen Datenspeicherung in der DDR."

Dr. D. B. G. Edwards, Electrical Engineering Laboratories, University of Manchester, England, "The design construction and use of a multi-track variable reluctance reading head in conjunction with digital computers."

T. Szentivanyi, Computing Centre of the Hungarian Academie of Sciences, Budapest, "Special Headconstructions for the Recording of Pulses."

Dr. L. Edelényi, Telephone Factory, Budapest, "Transient Phenomena by Magnetic Disk Memories with Special Track Selection."

Signal-to-Noise Ratio and Equalization of Magnetic-Tape Recording*

HANSWERNER PIEPLOW†, SENIOR MEMBER, IRE

Summary—Future tape recorders running at a speed of 1⅞ ips must achieve an upper frequency limit of 15 kc on a quarter track recording, with better SNR than presently obtained with commercial recorders. This means that the subjective quality of present full track machines at 15 ips must be maintained in spite of a fifty-fold increase of information density.

It is obvious that such an improvement cannot be obtained only by changes to the electronic circuitry and the like. In fact, the future requirement would be impossible to meet if improvements in tape and heads were not made. Such improvements must include new knowledge of head construction, tape-to-head contact problems, and optimum balancing of recording parameters. But once these improvements are effected, it is then necessary to check carefully if the most thorough use is being made of all available technical possibilities.

It will be shown that recording techniques have advanced to a stage where a change in the standardization characteristics is advantageous and even necessary, though they may be more complicated than the existing standards.

Procedure of the Investigation

Because the hum does not physically limit the recording technique and, if necessary, may be eliminated completely, we have to consider only the noise generating from the input stage of the recording amplifier, from the tape itself, and from the input stage of the replay amplifier. These noises are not correlated in any way, and each of them can be minimized without influencing the others. Since noise optimizing techniques in amplifier stages are in common use and since the absolute value of tape noise unfortunately does not depend very much on the bias level or on the constitution of the magnetic layer, there is no other chance to improve the SNR except by increasing the signal on the tape.

As trivial as this conclusion looks, it does not seem to be generally appreciated. For instance, in case of essential improvements of tape and heads, it is not necessary to decrease the pre-emphasis in order to match the existing standards,[1] but we must change the standards in order to increase the SNR.

Thus, with the basic method of standardization in mind we have to evaluate the admittable amount of preemphasis and must compare the results to the actual situation of technical art.

Method of Standardization

Since intensity, frequency response, and distortion of a recorded signal depend upon the individual properties of the recording machine and since these properties cannot be defined in general terms, standardization can be applied only to the tape itself, *i.e.*, to the level and to the frequency characteristic of the remanent magnetic surface flux of a recording. Once a standard tape is established, each machine can be adjusted to fit the standards.

If a magnetic tape is magnetized by a constant magnetic field, *i.e.*, by keeping the signal current constant in the recording head, the remanent surface flux on the tape is not constant, but decreases approximately exponentially with increasing frequency (Fig. 1). Now, the basic postulate of an international standardization

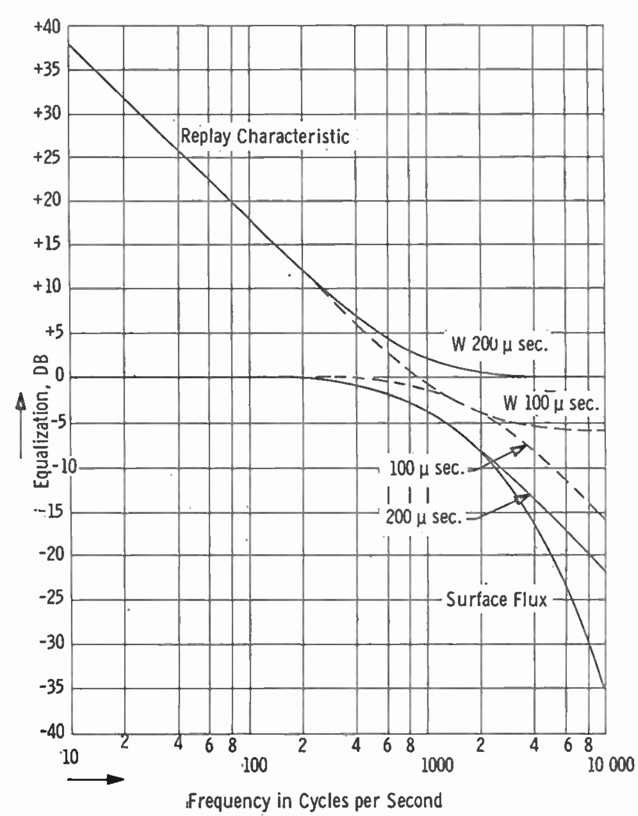

Fig. 1—Comparison of standard equalizations.
100 μsec, 200 μsec: Impedance characteristics of a resistor and capacitor in parallel with time constants of 100 μsec and 200 μsec, respectively.
W100 μsec, W200 μsec: Replay characteristics.

* Presented at the 1962 IRE International Convention, New York, N. Y.

† Grundig Radio Werke, Furth, West Germany.

[1] J. C. McKnight, "Some new data on frequency response of magnetic recorders for audio," *J. Audio Engrg. Soc.*, vol. 8, pp. 46–53, no. 3; 1960.

states that with a constant signal voltage at the input of the recording amplifier the flux vs frequency on tape has to vary in a predetermined manner, for example, like the impedance of a resistor and a capacitor in parallel with a time constant of 200 μsec.

As seen from Fig. 1, we need a certain pre-emphasis to meet this demand, coming up to 14 db at 10 kc with 200 μsec and to 20 db at 10 kc with 100 μsec. The smaller the value of the time constant, the higher the pre-emphasis required. Thus more signal is recorded on the tape, and a better SNR is achieved.

The Admittable Amount of Pre-emphasis

Of course, to avoid nonlinear distortions, a standardization cannot prescribe arbitrarily small time constants and it is necessary to make sure that the recording system will not be overloaded. Overloading occurs theoretically when the amount of pre-emphasis exceeds in any way the slope of the spectral peak distribution of music.

Unfortunately very different spectra are published, ranging from a decrease of 18 db at 10 kc[2,3] up to no decrease at all[4] in modern music, and in spite of the most ingenious considerations no reasonable method has been found to match pre-emphasis and frequency distribution of peak energy in music.

On the other hand there is no exact information about the probability of occurrence of a special event, *i.e.*, how frequently, for example, a 10-kc peak energy level will occur in all music. Therefore, following rigorously an orthodox line with regard to peak energy distribution and exclusing consequently any pre-emphasis, we may run the risk of recording 99 per cent or more of all musical playing time with an inferior quality, *i.e.*, with a smaller SNR than we could do, whilst recording 1 per cent or less of the whole musical playing time without hypothetical distortions in the upper frequency range whose audibility is not even proven.

For these reasons it looks more realistic to go back to the practical experience. In the last 10 years more than one million tape recorders have been produced in the Grundig factories with an average pre-emphasis of at least 15 db at 10 kc and without any objection to distortions. This fact is a very strong argument in favour of 15-db pre-emphasis at 10 kc.

The Actual Situation in Recording Art

Fig. 1 states the average situation of head and tape, used in home tape recorders at $3\frac{3}{4}$ ips in 1958. A more precisely defined measurement is shown in Figs. 2 and 3 where the a) curves represent the head EMF and the remanent surface flux, respectively vs frequency at $3\frac{3}{4}$ ips, the head being a combined recording and reproducing half track head demonstrating the statistically mean value of a large number of heads produced in 1959, the tape being the DIN—Bezugsband which is a German reference tape with magnetic properties guaranteed by the Physikalisch-Technische Bundesanstalt. As can be seen from Fig. 3, a 100-μsec equalization would require a pre-emphasis of 18 db at 10 kc which is a fairly high value, if compared to the conditions of the measurement. This high value has the main reason why up to now the majority of German and English tape recorders were equalized following the IEC recommendations with 200 μsec instead of the NAB standards with 100 μsec.

During 1960, as a consequence of very intense research work in the field of head design and tape manufacturing, essential improvements have been made. The b) curves in Figs. 2 and 3 represent the actual situation and can be compared directly to the a) curves with the only difference being that the b) curves refer to quarter track and to a tape speed of $1\frac{7}{8}$ ips. The very big progress

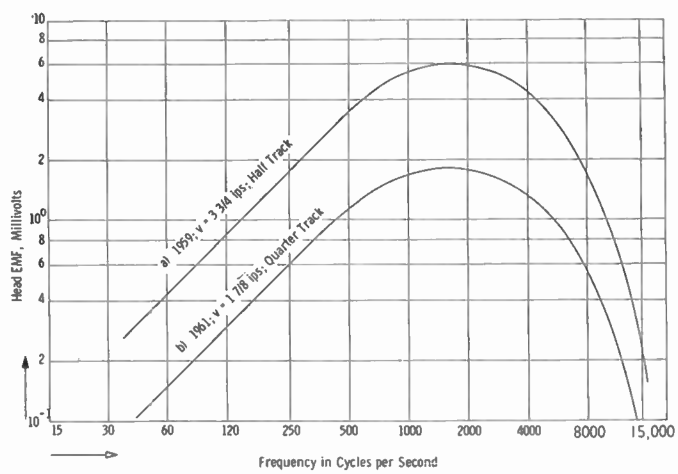

Fig. 2—Head EMF of a constant signal current recording with 4 per cent equalized distortion. a) 1959, combined half track head, Tape DIN-Bezugsband. b) 1961, separate recording and reproducing heads, quarter track, Tape LGS 26, batch no. 110211.

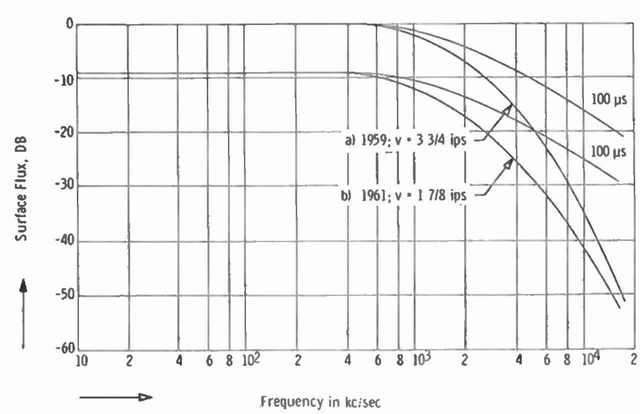

Fig. 3—Surface fluxes deduced from Fig. 2.

[2] J. P. Overley, "Energy distribution in music," IRE Trans. on Audio, vol. AU-4, pp. 120–123; September–October, 1956.

[3] R. H. Snyder and J. W. Havstad, "Equalization considerations in direct magnetic recording for audio purposes," 1956 IRE National Convention Record, pt. 7, pp. 134–141.

[4] J. G. McKnight, "The distribution of peak energy in recorded music, and its relation to magnetic recording systems," *J. Audio Engrg. Soc.*, vol. 7; April, 1959.

issuing from Figs. 2 and 3 does not only refer to the level, because other factors being equal a loss of 10 db in level results from the 3 to 1 reduction of track width alone, but refers particularly to the frequency response allowing now a 100-μsec equalization at $1\frac{7}{8}$ ips with a pre-emphasis of 16 db. Before this was impossible or at least risky even at $3\frac{3}{4}$ ips.[5]

Existing Proposals

It appears that the evolution of technical art runs faster than the standardization work of the different committees. The members of the IEC are still arguing about 100 or 120 or 140 μsec at $3\frac{3}{4}$ ips. Even the last German proposal for $1\frac{7}{8}$ ips[6] shows a surface flux vs frequency varying like the transfer characteristic of a network composed by two equal RC meshes, each of them with a time constant of 70 μsec. As can be seen from Fig. 4, the results of such an equalization would not be too satisfying:

1) The playback amplifier needs an equalization of 210 μsec, which is too much and should be avoided with respect to a good SNR.
2) The response of an ideal playback head needs a post-emphasis of 4.5 db at 10 kc.
3) By comparison to Fig. 3 the pre-emphasis while recording is only 5 db at 10 kc. This is much too small with respect to a good SNR.

New Equalization for Home Tape Recorders

The present state of art in Figs. 2 and 3 was achieved:

1) By the use of a very thin and, therefore, very pliable tape of high coercivity. The tape must have a very smooth, highly polished surface as obtained for example with LGS 26 tape.
2) By the use of separate heads for recording and reproducing, which are particularly designed to have extremely straight and extremely sharp gap edges.
3) By carefully choosing the bias of the recording head in order to get the best compromise between level, distortions and frequency response. This bias does not necessarily coincide with that giving maximum output at 1 kc. The better the tape matches high-frequency performance, the smaller the difference is in bias.

It has been proved that the state of the art, demonstrated by Figs. 2 and 3, can be maintained in mass production. It seems that the standardization proposals existing until now have not taken this performance into account. Therefore, we have introduced a 100-μsec equalization at $1\frac{7}{8}$ ips in our machines and, additionally, a pre-emphasis at the lower end of the frequency range corresponding to the impedance of a resistor and a capacitor in series with a time constant of 1591.5 μsec.

[5] J. G. McKnight, "The effect of bias amplitude on response at very short wavelengths," *J. Audio Engrg. Soc.*, vol. 9, pp. 98–102; no. 2, 1961.
[6] DIN 45513; December, 1960.

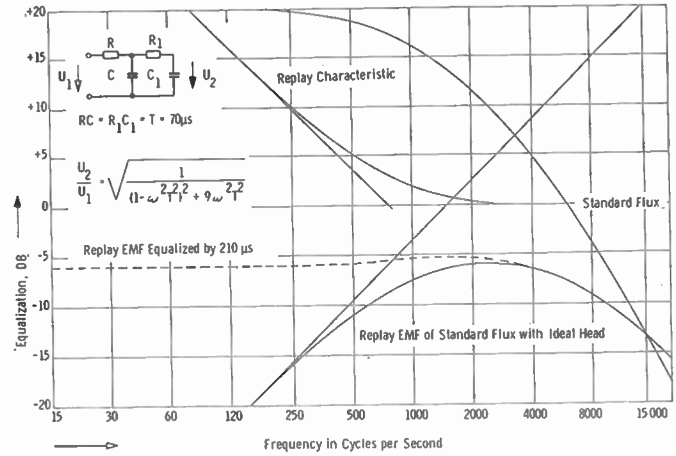

Fig. 4—DIN proposal 1961 for standard equalization at $1\frac{7}{8}$ ips.

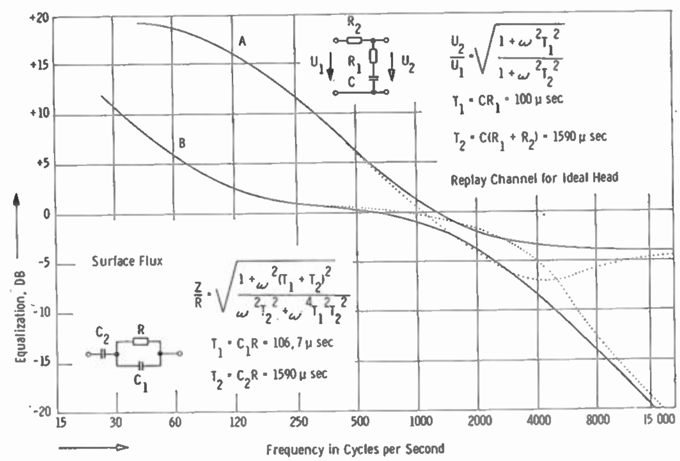

Fig. 5—Grundig equalization for $1\frac{7}{8}$ ips. A) Replay channel equalization. B) Tape surface flux. (dotted lines: with additional noise suppression equalization)

The time constant of 1591.5 μsec is half the value of the actual NAB equalization which is felt to be too high because, if any pre-emphasis at all is used at the lower end of the frequency range, it should have some perceptible influence at 180 cps, *i.e.*, at the third harmonic of the power frequency. Of course, this pre-emphasis has nothing to do with the physics of the magnetic recording, but it is introduced for practical reasons, because the machines become cheaper.

The complete new equalization for $1\frac{7}{8}$ ips, including analytical formulas of its definition, is shown in Fig. 5.

Referring to $3\frac{3}{4}$ ips, it is not favourable to change the time constant. Better SNR may result in shifting the bias to the 1-kc maximum point. In this way the continuity with the NAB equilization is maintained at the same time. At $7\frac{1}{2}$ ips the time constant has to be cut down to 50 μsec.

Noise Measurement

With regard to SNR it is necessary to define the signal and the noise in order to get a realistic figure of merit to judge the quality of a machine or of an equalization system.

Unfortunately, signal and noise measurements are not standardized. Of course, the signal measurement or more precisely, the maximum signal measurement is very clear and may be achieved by measuring the output at a given frequency, say 333 or 1000 cps, at a given distortion, for example 5 per cent.[7] However, it is not at all correct to measure the noise by a linear instrument independent from frequency in the whole frequency range. In addition, it is not correct to suppress the hum by a high pass filter. In order to get an objective measure of a subjective noise impression, it is necessary to take account of[8,9] the ear sensitivity vs frequency and vs loudness and the spectral noise energy distribution, which is essentially "white" and which, therefore, has an increasing energy content per octave with increasing frequency. For this combination of ear sensitivity and energy distribution we hear mostly a hiss, although the hum may have a higher amplitude.

To meet the demand for the objective measurement of a subjective noise impression, the CCIF[10] has published a weighting characteristic (Fig. 6) which is used by all European broadcasting stations. It is enigmatical why this widely used method has not yet officially entered the tape-recording field and seems to be unknown even to professional tape-recorder people.[8,9,11] There is no doubt that using the weighting characteristic of Fig. 6 which, for example, is built into the Siemens Gerauschspannungsmesser Rel 3 U 33 the SNR will become a realistic and very distinct figure of merit.

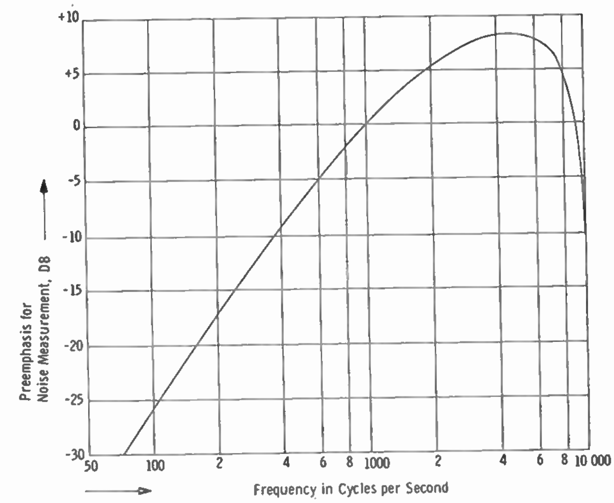

Fig. 6—Weighting characteristic for noise measurement of CCIF.

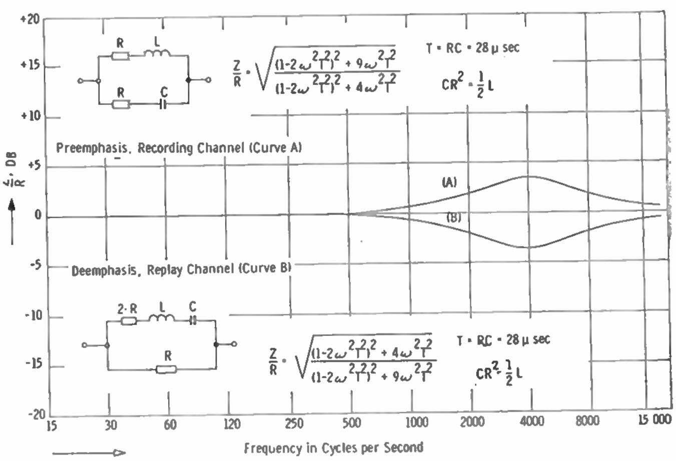

Fig. 7—Proposed noise suppression equalization.

Noise Suppression Equalization

From Fig. 6 we can deduce easily how to get the best possible SNR. If the most troublesome noise frequencies are found to be between 3 kc and 6 kc, the signal has to be pre-emphasized mostly in this frequency region.[9] No reason can be seen as to why this method should not work at lower tape speeds,[8] and listening tests showed clearly a subjective improvement in noise impression without appreciable overloading even at $1\frac{7}{8}$ ips.

A new tentative proposal for standardization of a noise suppression equalization is given in Fig. 7. It has practicable values and can be defined analytically in a very closed form. If applied to normal recording and reproducing channels, it may be simply superposed (dotted lines in Fig. 5).

This type of equilization looks somewhat complicated and, at the present, may not be necessary in usual home recording. But it brings up SNR by 3 to 4 db and may be very useful for prerecorded music on tape of high information density.

Conclusions

As long as home tape recording is used just for fun or in rare hi-fi equipment, it makes no difference whether tape is wasted or not. But in case tape is used as a high quality sound carrier and must undergo competition of disk recording, there is no doubt that high quality has to be achieved with high information densities, *i.e.*, with small tracks and low speeds.

The broad research work, which was initiated by Grundig Radio Werke during 1960, has clearly shown that the improvements possible in the tape and head physics and technology bring tape recording into a new region no longer adequately described by the existing standards for equilization, be they NAB or DIN or IEC.

New equalization proposals were worked out and have been introduced into mass-produced tape recorders with the result that, with a quarter track at $1\frac{7}{8}$ ips, the frequency response is flat up to 12 kc within normal tolerances and that SNR is better than 47 db.

An additional noise suppression equalization could be defined which may be useful for prerecorded music.

[7] DIN 45511; March, 1955.
[8] J. G. McKnight, "Signal-to-noise problems and a new equalization for magnetic recording of music," *J. Audio Engrg. Soc.*, vol. 7, pp. 5–12; 1959.
[9] A. A. Goldberg and E. L. Torick, "A new equalization characteristic for master tape recording," *J. Audio Engrg. Soc.*, Preprint No. 115; October, 1959.
[10] Greenbrook, CCIF, Geneva, Switz., vol. 4; 1956.
[11] DIN 45510; July, 1955; DIN 45511; March, 1955.

Design Aspects of FM Stereo Tuners and Adaptors*

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Summary—**The stereophonic composite signal may be decoded in adaptor circuitry by time multiplex or sum-and-difference approaches. To obtain suppression of SCA (background music) signals while maintaining adequate separation of the two stereophonic audio signals double, single, or residual sideband demodulation of the stereo subcarrier can be employed. Satisfactory reception of FM stereo signals requires tuner circuitry of higher performance than for equal monophonic performance. Calculated and measured stereophonic separation, distortion, and signal-to-noise ratio are shown with major causes of these performance aspects.**

THE SPECIFICATIONS for compatible stereophonic broadcasting state that the main channel be modulated by the sum of the left and right audio channels. A subcarrier of 38 kc also modulates the FM transmitter and is in turn amplitude-modulated by the difference of the left and right audio channels. The 38-kc subcarrier in turn is suppressed, and a 19-kc pilot carrier is transmitted at 10 per cent modulation for synchronization of the stereo demodulator. All of these signals are available at the output of an FM detector for the stereo demodulator.

Two basic types of stereo demodulators can be constructed. The sum-and-difference type of stereo demodulator selects the subchannel signal by means of a band-pass filter and uses an AM detector with a reinserted 38-kc carrier for demodulation of the difference channel signal. The difference channel modulation then is added to the main channel to recover the left channel output. If the phase of the difference channel signal is reversed 180 degrees and added to the main channel, the right channel is obtained.

Examination of the composite signal also shows that this composite signal may also be described as a time-division multiplex signal with the higher harmonics due to switching removed by a 53-kc low-pass filter. Stereo demodulator circuitry can be constructed on the same basis. Here, the left and right channel outputs are switched at a 38-kc rate successively to the composite signal. By this means, the composite signal is decoded.

As shown above, the difference channel signal is double sideband suppressed carrier amplitude modulation, and either one or both sidebands carry the full stereo information. Since the FCC specifications also permit the station operator to broadcast SCA signals at frequencies above 53 kc, it is important that the stereo demodulator circuitry be insensitive to these high frequencies. The necessity of maintaining matching amplitude and phase response in the sum and difference channels require a phase linear low-pass filter at the input of the stereo demodulator. This may attenuate frequencies below 53 kc. However, stereo demodulation can still be accomplished because single sideband, double sideband and vestigial sideband detection are possible. Here, the loss in subcarrier signal can be corrected by increasing the differential gain between the two audio amplifiers following the stereo demodulator.

In order to maintain the separation between left and right channels of the order of 30 db, it is necessary that the differential phase response of the main and subchannels be maintained to approximately 3.5°, and the differential amplitude to approximately 0.5 db. Since correction for both of these factors is possible ahead or following stereo demodulator circuitry, results shown in Fig. 1 may be typical of a high performance stereo tuner. In contrast with this, the frequency response of monophonic tuners is generally maintained only to no better than ±1 db between 50 and 15,000 cps.

To obtain good stereophonic performance, it is also necessary that the distortion of the audio output signal at the left and right stereo demodulator outputs be kept to a reasonably low value. If it is assumed that the distortion is created entirely in the circuits ahead of the deemphasis network, the measured over-all distortion falls with an increase in frequency in normal monophonic service and the harmonics of the higher audio frequencies will be inaudible. Even if the distortion of the signal rises with frequency, distortion is held to a relatively constant value between 50 and 15,000 cps.

In stereophonic service, the stereo demodulator is sensitive to frequencies between 23 and 53 kilocycles. For example, harmonics of the main channel may fall into the subchannel and will cause beats. Also, the normal method of measuring harmonic distortion is not quite applicable because the presence of any desired supersonic signals may show up as distortion. Only undesired audio frequency components below 15 kc are therefore regarded as distortion. Fig. 2 shows the amounts of distortion calculated as function of frequency when only the main channel or the subchannel are fully modulated and no deemphasis network is used. It can be seen here that for a certain range of audio frequencies no distortion will be measured. The lower cutoff frequency is the second, third or fourth subhar-

* Received by the PGA, October 25, 1961. This paper was presented at the Northeast Electronics Research and Engineering Meeting, Boston, Mass., November 14, 1961, and appeared in the *NEREM Record*.

† H. H. Scott, Inc., Maynard, Mass.

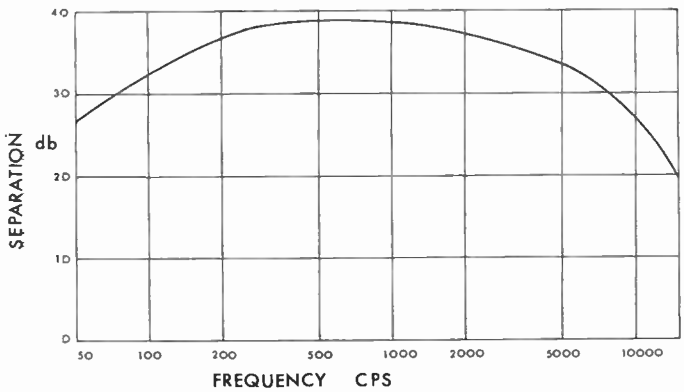


Fig. 1—Curves illustrating typical results of a stereo tuner.

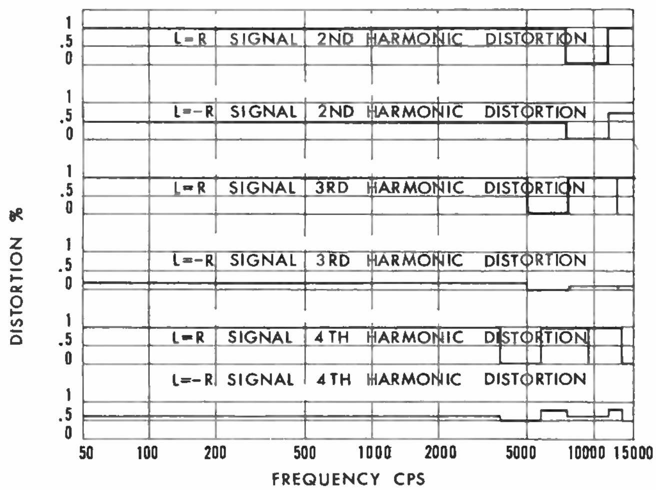


Fig. 2—Distortion at L and R output with no deemphasis, 15-kc low-pass filter and 1 per cent composite signal distortion.

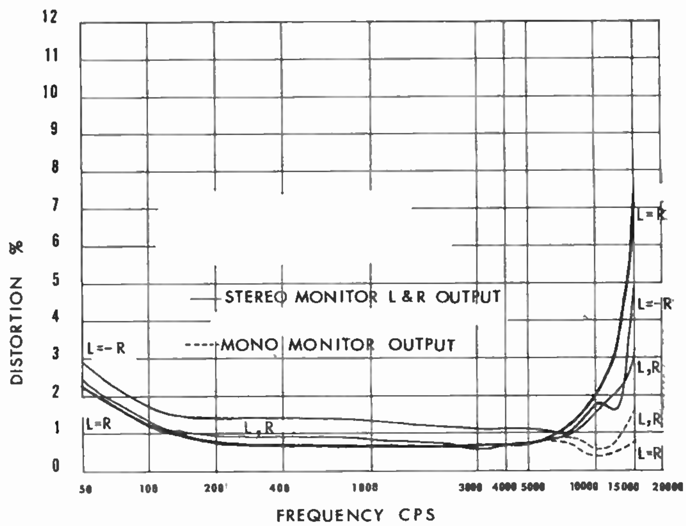


Fig. 4—Distortion vs frequency curves at 100 per cent modulation measured at the NSRC transmitter site in Pittsburgh.

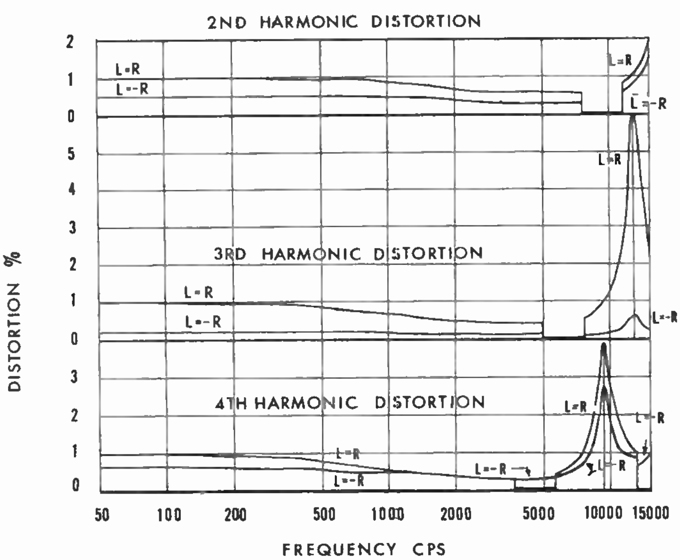


Fig. 3—Distortion at L and R output with a 15-kc low-pass filter and 1 per cent composite signal distortion, but with 75-μsec deemphasis.

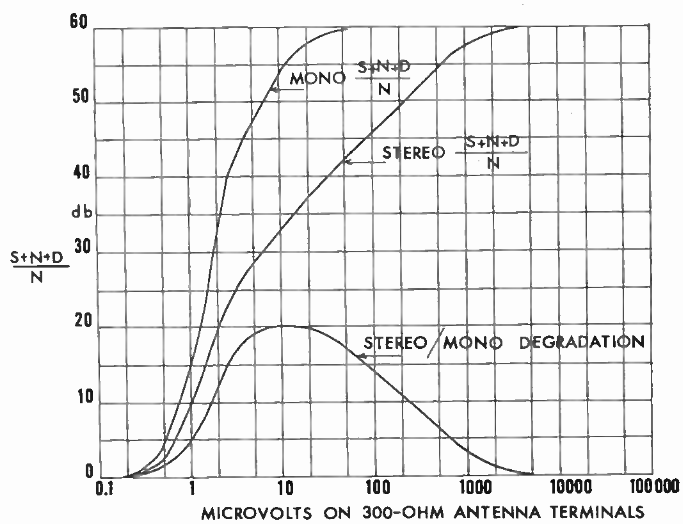


Fig. 5—Signal-to-Noise Ratio of tuner with a stereo adapter.

monic of 15 kc, and the higher cutoff is the first, second, or third subharmonic of 23 kc. The higher frequencies result in beat tones only. Fig. 3 shows the same calculations but with 75-μsec deemphasis. Here, apparent distortion falls with frequency up to the subharmonic of the 15-kc filter. At higher frequencies, the desired signal is attenuated by the deemphasis network, whereas the beat tones are only partially attenuated. The presence of deemphasis, however, causes apparent distortion to be twice as much for second harmonic distortion, higher by a factor of 6 for third harmonic distortion, and 4 for fourth harmonic distortion.

Results very similar to those calculated were observed at the NSRC field tests in Pittsburgh. Here, distortion measured at the transmitter is shown for both a monophonic and a stereophonic monitor. The curves of Fig. 4 show a remarkable similarity to those in Fig. 3. The continued rise of distortion with frequency may be presumed to be caused by a rise of composite signal distortion.

In design of stereophonic tuners, therefore, particular care has to be exercised to keep high-frequency distortion at a minimum.

Calculations also indicate that the signal-to-noise ratio at the stereophonic outputs is approximately 20 db poorer than the SNR of the corresponding main channel only. Measurements agree closely with this prediction as shown in Fig. 5. Here, the ultimate SNR was limited primarily by hum and residual noise of the generating equipment. If the signal-to-hiss ratio were measured, the ultimate signal-to-hiss ratio would be in the order of 90 db for the monophonic signal and about 70 db for the stereophonic signal. Even though the art of designing stereophonic tuners is still young, the results at the present time show that tuner performance compares favorably with broadcast performance.

A Versatile Phonograph Preamplifier Equalizer*

HAROLD FRISTOE†, SENIOR MEMBER, IRE

Summary—A phonograph preamplifier-equalizer system has been developed which permits continuous equalization adjustment over a wide frequency range. The playback contour can therefore be "fitted" to accommodate any recording characteristic normally encountered in practice.

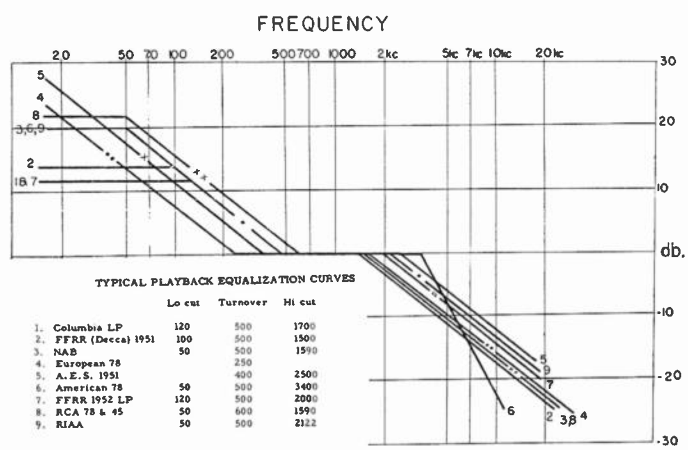
THE REPRODUCTION of music recorded on disks requires a preamplifier system with special response characteristics. The preamplifier-equalizer normally serves a dual purpose. First, the preamplifier must furnish a specified amount of gain. Second, equalizing networks are included to compensate for the recording characteristic imposed on the disk by the manufacturer as well as the inherent conversion characteristics of the particular pickup employed. Equalization circuits are normally arranged so that only predetermined fixed equalization characteristics can be selected by step switching arrangements.

Historically, recommended playback equalization data has been difficult to obtain and has not always been reliable. Many of the older recordings deviate enough from "typical" recommended characteristics that before reproduction sounds right the listener needs equalization freedom beyond that offered by preset compensating networks. This immediately suggests that equalizers need far more flexibility than that possible with discrete step selection. Present recordings using the RIAA curve are quite consistent compared to older recordings, but often record collections still include pre-RIAA disks.

Two alternatives exist for proper equalization: a fixed reference curve may be built into the preamplifier and an elaborately designed set of base and treble controls then allow adjustable compensation; or the equalizer system itself can be so arranged that continuous variation around a particular recording characteristic will be available. At the cost of adding one more article to the mounting list, a phonograph equilizer-preamplifier will be described which has the unique flexibility of continuous contour adjustment over a rather wide frequency range. This flexibility results in an increased complexity of controls, but there is a group of serious audiophiles who are willing to tolerate these adjustments to that reproduction will be completely satisfying.

A careful analysis of various recording characteristics, Fig. 1, dictates the minimum range of control necessary to satisfy "typical" recording characteristics. By extending the equalization range sufficiently beyond this minimum a reasonable amount of tone control action will also permit corrections for deficiencies which may exist in the system and the acoustic environment. It is therefore desirable that the peramplifier-equalizer have the following characteristics:

1) Continuous control of turnover frequency over the range of 200 cps to 1600 cps.
2) True 6-db-per-octave rise below the turnover frequency.
3) Continuous low-frequency rolloff below 200 cps.
4) Continuous high-frequency rolloff adjustment of 6-db-per-octave above 1.2 kc.
5) Step selection of an additional 6-db-per-octave high-frequency cut at 1.6 kc, 3.4 kc, and 6 kc.
6) Independence of all equalizer adjustments.
7) Low noise circuitry.
8) Utilize the magnet pickup.
9) An output voltage of approximately 0.5 volt available to drive a remotely connected power amplifier.
10) Output impedance sufficiently low to drive a long cable which feeds the power amplifier.



	Lo cut	Turnover	Hi cut
1. Columbia LP	120	500	1700
2. FFRR (Decca) 1951	100	500	1500
3. NAB	50	500	1590
4. European 78		250	
5. A.E.S. 1951		400	2500
6. American 78	50	500	3400
7. FFRR 1952 LP	120	500	2000
8. RCA 78 & 45	50	600	1590
9. RIAA	50	500	2122

Fig. 1.

Placing the preamplifier adjacent to the turn table has several inherent advantages. A long low-level cable which ordinarily connects the pickup to the preamplifier is eliminated. Such cables may appear as a source of hum and introduce capacity loading of the pickup which results in a resonant hump in the reproduction characteristic of a magnet pickup.[1] Using vacuum tubes in a preamplifier located at the turn-table introduces several objectionable problems. Tubes are subject to vibration,

* Received by the PGA, December 26, 1961.
† School of Electrical Engineering, Oklahoma State University, Stillwater, Okla.

[1] N. Pickering, "Effect of load impedance on magnetic pickup response," *Audio Engrg.*, vol. 37, pp. 19–20, 26; March, 1953.

which results in microphonic noises. The problems of supplying power to the tube circuits and adequate ventilation to remove the heat generated in the tubes can offset the advantage of a short connecting cable. Tubes and tube circuits are also subject to hum voltages.

These objections are eliminated by utilizing transistors in the amplifier circuits. Installation of the preamplifier adjacent to the turn table is then practical. The power demand of the amplifier is low enough that mercury batteries will last well over a year under normal operation. Furthermore, a relay may be arranged so that the batteries are turned off at all times except while the turn table motor is energized, thus conserving battery capacity.

Circuit Description

The block diagram of the preamplifier-equalizer is shown in Fig. 2. A General Electric Type VR II magnetic pickup was coupled to the input of the preamplifier through a short interconnecting shielded cable. A wealth of technical information is available for this pickup. Consequently, exact equalization can be introduced in the preamplifier so that the output signal from the pickup will be properly proportioned to the exciting force applied at the stylus tip. Resistance loading[2] of the magnetic pickup fulfills the requirement for high quality performance as well as design simplicity. Resistance loading is very desirable when driving into a transistor stage since the recorded high-frequency preemphasis is cut down. The reduced high-frequency level of the signal applied to the transistor base reduces the possibility of overdriving any of the succeeding stages. Furthermore, high-frequency equilization is readily achieved as the load resistance is adjusted. Varying this resistance from approximately 50 K ohms to as low as 2.2 K ohms results in a continuous variation of the high-frequency rolloff extending from well above 20,000 cps to about 1200 cps.

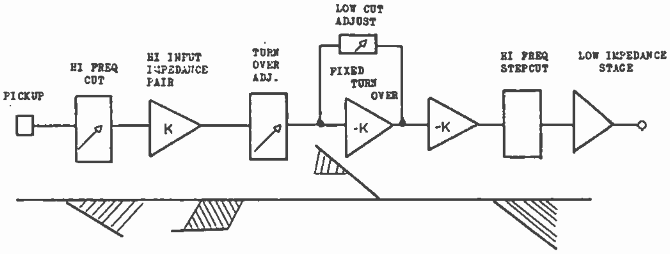


Fig. 2—Basic amplifier showing equalization adjustments.

There are two possible connections whereby satisfactory de-emphasis may be introduced at the input through resistance loading of the pickup. First, the variable resistance may be placed in series between the pickup and the input of the first grounded emitter amplifier stage (Appendix I). Using this connection, it is discovered that the low-frequency base current supplied to the transistor will vary approximately 19 db referred to mid-frequencies when the control is adjusted over the frequency limits specified. This type of connection would result in an undesirable variation of average output signals as the equalization control is adjusted. Second, a parallel connection can be used (Appendix II). In this case, low-frequency current driving the base will show approximately 2 db variations as the control is varied throughout its limits as long as the dynamic input impedance of this stage is made high. Consequently, there will be only minor reaction between the high-frequency adjustment and the other adjustments in the equalizer circuit. The schematic of Fig. 7 (b) (Appendix II) shows a network consisting of R and R_s paralleled by Z_{in} of the first transistor stage. Varying resistor R_s causes the dynamic load on the pickup to change from about 2 K ohms to 35 K ohms, thus varying the upper corner frequency continuously from 20 kc to 1.2 kc. In order for the transistor to present a high dynamic input impedance as required by the parallel connection, the bootstrap arrangement[3] shown in Fig. 3 is employed. The voltage gain of this stage is quite low and is followed by a second stage which introduces normal power gain.

The second stage is followed by the low-frequency equalization network and amplifier which permits turnover adjustment, as well as adjustable low-frequency rolloff. Variable turnover frequency can be obtained by applying a selective feedback loop around an amplifier as shown in Fig. 8 (Appendix II-A). In order to vary the turnover frequency, the resistance of the feedback loop can be varied. In turn, the stage gain above the turnover frequency will vary appreciably, Fig. 8 (Appendix III). This method of turnover adjustment was rejected since independent control action was desirable.

A novel method for obtaining variable turnover frequency was employed (Appendix IV). A frequency sensitive step network, external to a fixed turnover amplifier, provides a convenient method for adjusting the turnover frequency with no appreciable effect upon the high-frequency gain. The fixed turnover frequency was set at approximately 2000 cycles with a 6-db-per-octave rise. At least 40 db of current gain is required in this stage if true 6-db-per-octave rise is maintained into the low-frequency region. This is possible by using a transistor with an h_{fe} of 200 or more. A low impedance feedback loop connected across the stage gives the characteristic shown in Fig. 9 (Appendix IV). The input impedance of a stage of this type becomes extremely low above the turnover frequency, so that the approximate analysis given in Appendix IV is considered valid.

[2] N. S. Cromwell, "Letters," *Audio Engrg.*, vol. 38, pp. 8, 25, September, 1954. C. F. Hempstead and H. Barhydt, "Accurate design of phono equalizers," *Audio Engrg.*, vol. 38, pp. 24–25, 65; September, 1954.

[3] W. D. Roehr, "Characteristics of degenerative amplifiers having a base emmitter shunt impedance," IRE Trans. on Audio, vol. AU-7, pp. 165–169; November–December, 1959.

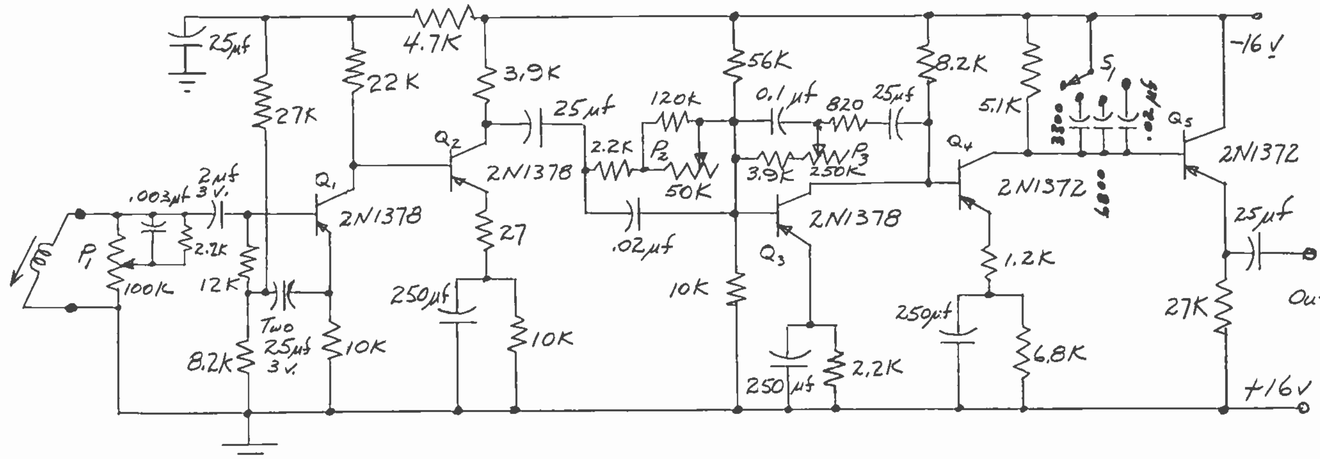


Fig. 3—Circuit diagram of preamplifier-equalizer. P_1 = high-frequency control, P_2 = turnover control, P_3 = bass cut control, S_1 = high-frequency step control.

In order to adjust the turnover frequency to something other than that set by the feedback loop, a step network is connected between the output of the second stage and the input of the turnover stage. This will produce a frequency selective loss below the fixed turnover frequency. The desired loss curve required to shift the turnover frequency is shown in Fig. 9. Above the turnover frequency no loss is desired, while below the new turnover frequency a fixed loss is introduced. This is achieved by limiting the driving current into the transistor base by utilizing the frequency selective current divider. A capacitor placed across the base current limiting resistor produces this selective effect. The circuit components were selected so the turnover frequencies were adjustable from below 200 cps to above 1500 cps.

Low-frequency rolloff is achieved by shunting the capacitor in the feedback network with resistance, thus limiting the stage gain below this corner frequency by the resistive feedback placed around the amplifier stage.

Stage four is a conventional common emitter amplifier in which current feedback has been introduced in order to minimize loading effects on the third stage. A direct coupled, common collector circuit then follows. Thus, a low driving impedance is presented to the loading capacity of the terminating line.

A switch is arranged between the third and fourth stages so that various capacitors can be introduced. These capacitors were chosen so that high-frequency rolloff is introduced at specified fixed corner frequencies. The de-emphasis introduced at this point is completely independent of and additive to that introduced by resistance loading of the pickup. This additional rolloff may be employed to advantage when noisy records are being reproduced.

While temperature stability is normally not a severe problem in a unit of this type, it was considered desirable to maintain a high degree of collector current stability. The first transistor pair was arranged to have a current stability factor of about two, and the following transistor triplet was designed to have a current stability factor of approximately four. Employing such low values of current stability results in not only freedom from temperature effects, but typical parameter variations encountered among transistors will have only a minor effect on the operating point established for each of the stages. This is especially important in direct coupled, doublet, and triplet connections. AC degeneration is applied so that ac stability as well as dc stability will permit transistor interchange with little effect upon the equalizing characteristics or the total gain.

The circuit developed to incorporate the equalization techniques as discussed is shown in Fig. 3. Components were assembled on a copper clad insulating board which had been staked with terminals conveniently located. All controls were mounted on a 2″×7½″ aluminum plate attached to the component board. The completed unit was only 2½″ deep.

Transistor Q_1 was selected for low-noise properties as well as a high h_{fe}. Q_3 should also be chosen to have a high h_{fe}. When all five transistors had lower limit h_{fe} values, the amplifier showed about a 3-db drop in output as compared to the case when they were chosen to have normal h_{fe} values. The variable equalizing potentiometers were standard logarithmic taper units connected as shown on the circuit diagram. The controls were calibrated in terms of the associated corner frequencies. The tapered controls resulted in rather open, easily defined calibration markings. The controls can be calibrated by using the charts as shown in Fig. 4. If greater accuracy is desired, a generator can be used to apply a test signal to the input loop through a very low resistance placed in series with the pickup. As the frequency of the constant voltage generator is swept across the band, the output of the preamp can be monitored. An interpretation of the output voltage contour then specifies the corner frequency setting for the various control positions. Calibration points can also be verified by standard frequency test record playback measurements.

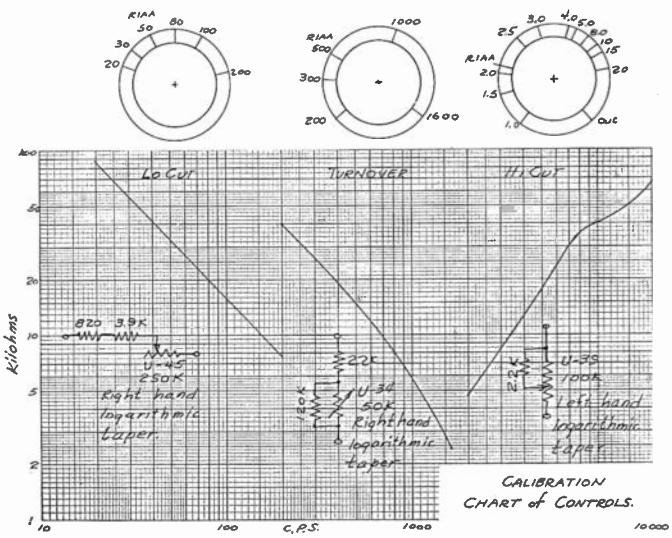


Fig. 4.

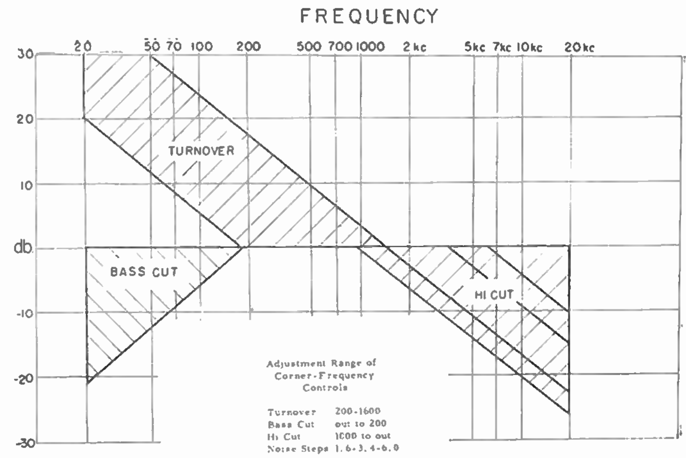


Fig. 5.

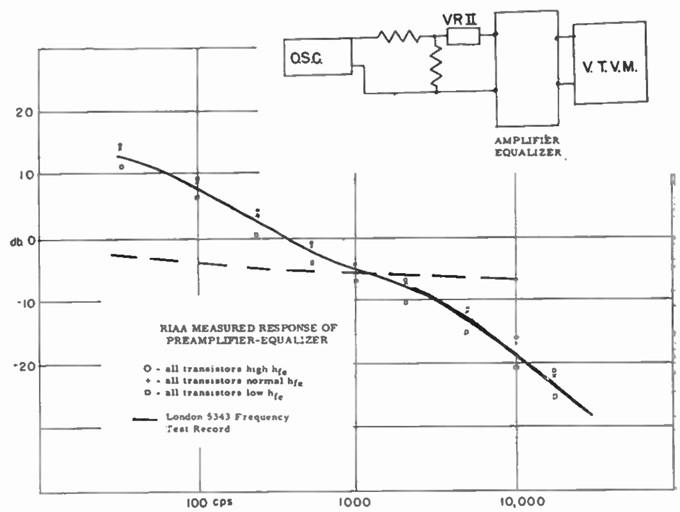


Fig. 6.

Performance

The equalizer-preamplifier design presented here meets the requirement of equalization flexibility and displays the following characteristics: mid frequency gain of 34 db; required input for 0.5-v output at 1000 cps is 10 mv with 0.25 per cent harmonic distortion; will accept an input of 20 mv before clipping occurs; separate controls independently vary the turnover frequency at a 6-db-per-octave rolloff with the corner frequency varying from 200 cycles to 1500 cps; a low-frequency 6-db-per-octave rolloff up to 200 cps; a high-frequency 6-db-per-octave rolloff from 1.2 kc to 20 kc; a four position switch selects an additional 6-db-per-octave cut at 1.6 kc, 3.5 kc, and 6 kc; thus combinations of the two high-frequency controls can reduce background noise associated with undesirable noisy recordings; self-generated noise is approximately 65 db below 1-v output at 1000 cps with the equalizer controls set for flat frequency response; hum problems are limited to the inductive pickup in the cartridge; the low output impedance of the emitter follower allows the amplifier to be used adjacent to the turn table and still be quite remote from the main amplifier and speakers. If the interconnecting cable capacity is assumed to be in the order of 50 pf, it will have little effect on the frequency response up to 100 ft. A shunt capacity of 50,000 pf on the output will attenuate 20 kc only 3 db. A 600-ohm resistive load will attenuate the output 3 db at all frequencies in the response range, provided the output coupling is increased to 250 μf or more. Thus the amplifier can be fed directly into a high impedance amplifier or into an unbalanced 600-ohm line. Transistors are quite insensitive to normal vibrations so the amplifier can be conveniently mounted beside the turn table without cushion mountings being required to control microphonics. The range of equalization is shown in Fig. 5.

Measured response of the amplifier is shown in Fig. 6. The equalizer controls had been set for the RIAA recording curve. The greatest deviation observed was less than one db. Base and treble boost or cut can be readily introduced by modifying the equalizer adjustments.

Appendix I

Consider a variable resistor connected in series with the magnetic pickup as shown in Fig 7(a). R_{load} is made up of the input resistance (h_{ie}) of a transistor in series with a variable resistor. As this resistor is varied the high-frequency corner-frequency is varied. The circuit current which would flow into the base of the transistor is:

$$I_B = E_S\left[\frac{1}{R_{\text{load}} + R + pL}\right] \tag{1}$$

$$= \frac{E_S}{(R_{\text{load}} + R)}\left[\frac{1}{1 + p\dfrac{L}{(R_{\text{load}} + R)}}\right]$$

$$R_{\text{load}} = R_S + \left[h_{ie} - \frac{h_{re}h_{fe}}{h_{oe} + Y_L}\right]. \tag{2}$$

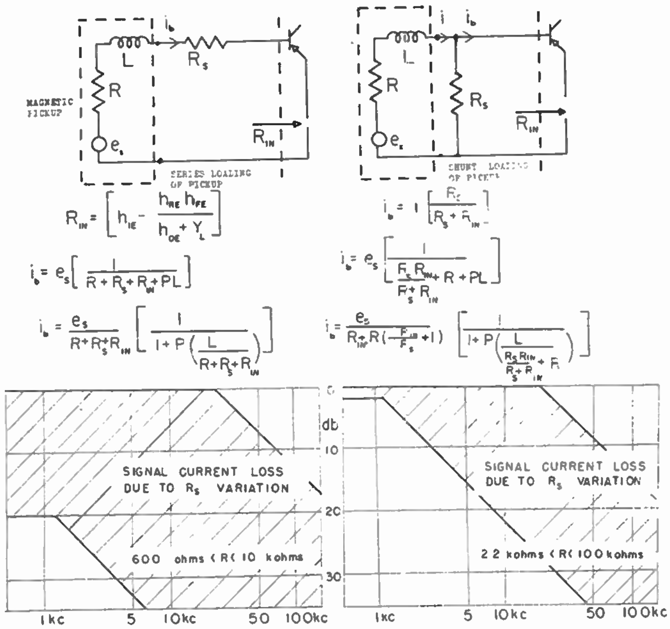


Fig. 7—(a) Series loaded pickup analysis.
(b) Shunt loading of pickup.

With the highest corner frequency set at 20 kc, R must be approximately 10 K ohms and with the lowest corner frequency set a 1.1 kc, R will have a value of approximately 600 ohms. The asymptotic response of the circuit for these two limits will compare as shown in Fig. 7(a). In the low-frequency region there will be a drop in the base current of 18.9 db referred to the 20 kc adjustment.

Appendix II

Using a transitor connection with dynamic input impedance paralled by the pickup loading resistor as shown in Fig. 7(b), the circuit will be analyzed. The current flowing into the transistor base will be written:

$$I_B = I\left[\frac{R_{\text{var}}}{R_{\text{var}} + R_{\text{in}}}\right] \tag{3}$$

but

$$I = E_S\left\{\frac{1}{\dfrac{R_{\text{var}}R_{\text{in}}}{R_{\text{var}} + R_{\text{in}}} + R_p + pL}\right\}. \tag{4}$$

Substituting four into three and simplifying gives the following expression:

$$I_B = \frac{E_S}{R_{\text{in}} + R_p\left(1 + \dfrac{R_{\text{in}}}{R_{\text{var}}}\right)} \cdot \left\{\frac{1}{1 + p\left\{\dfrac{L}{\dfrac{R_{\text{var}}R_{\text{in}}}{R_{\text{var}} + R_{\text{in}}} + R_p}\right\}}\right\}. \tag{5}$$

When R_{var} is changed from 100 K ohms to 2.2 K ohms the corner frequencies will again move from 20 kc to 1.1 kc, but there will only be a change of 2 db in base driving current in the frequency range below the corner frequency.

Appendix III

A frequency selective feedback network connected around a transistor amplified stage is shown in Fig 8.

The general expression for the current amplification of the stage can be written as:

$$K_i = \frac{h_{fe}Y_L}{h_{oe} + Y_L} \cdot \left\{\frac{1 + pC_F\left(R_F - \dfrac{h_{ie}}{h_{fe}}\right)}{1 + pC_F\left(R_F + h_{ie} + \dfrac{(1 - h_{re})(1 + h_{fe})}{h_{oe} + Y_L}\right)}\right\}.$$

At sufficiently low frequencies the term in the brackets reduces to unity so the low-frequency current gain of the stage approaches

$$K_{i\text{ low}} = \frac{h_{fe}Y_L}{h_{oe} + Y_L}$$

and is independent of R_F.

When the signal frequency becomes sufficiently high the current gain will be finally limited by R_F and will be expressed as:

$$K_{i\text{ high}} = \frac{(h_{fe}R_F - h_{ie})Y_L}{(h_{ie} + R_F)(h_{oe} + Y_L) + (1 - h_{re})(1 + h_{fe})}.$$

The two corner frequencies of the gain expression are then

$$f_1 = \frac{1}{2\pi C_F\left(R_F + h_{ie} + \dfrac{(1 - h_{re})(1 + h_{fe})}{h_{oe} + Y_L}\right)}$$

$$f_2 = \frac{1}{2\pi C_F\left(R_F - \dfrac{h_{ie}}{h_{fe}}\right)},$$

and the step height is

$$S = h_{fe}\left\{\frac{h_{ie} + R_F + \dfrac{(1 - h_{re})(1 + h_{fe})}{h_{oe} + Y_L}}{h_{fe}R_F - h_{ie}}\right\}.$$

As R_F is varied the turnover frequency is changed as well as the step height of the frequency contour. Consequently the high-frequency gain varies relative to the lower-frequency gain limit so that the average reproduced signal is not independent of R_F. If the feedback resistor is changed so the turnover frequency moves

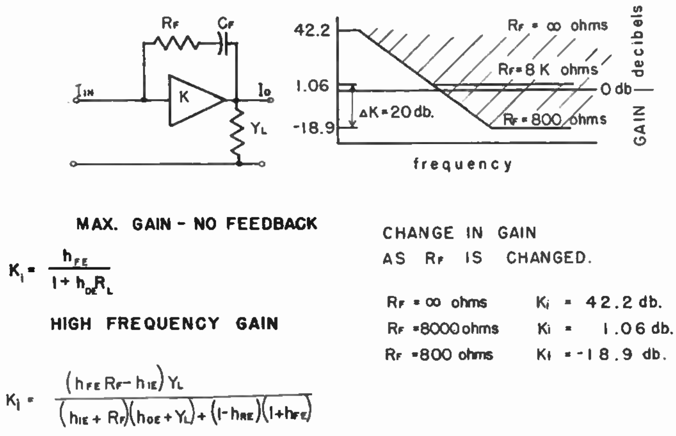


Fig. 8—Amplifier equations showing gain change caused by changing R_f of feedback network.

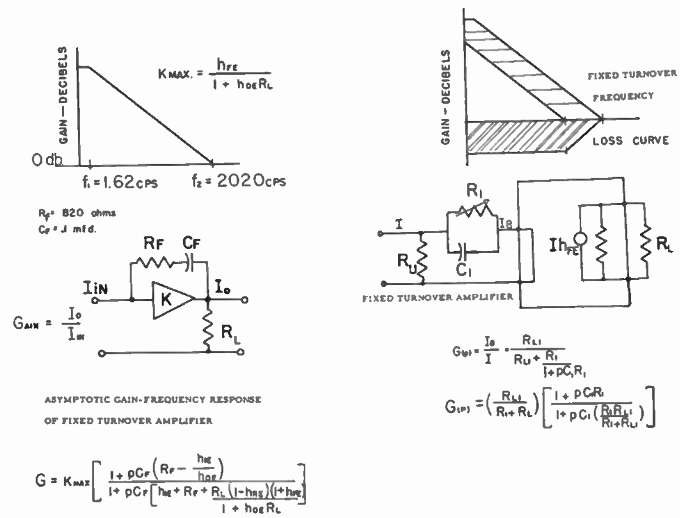


Fig. 9—Approximate equations and asymptotic response of adjustable step circuit to vary the turnover frequency.

from 200 cps to 1.6 kc the high-frequency gain will change about 38 db. Varying the turnover frequency would thus be impractical if mid-frequency gain is to remain independent of contour adjustment.

Appendix IV

The gain of the variable turnover stage can be made essentially independent of the turnover adjustment by combining a fixed feedback amplifier with a network which controls the base current drive. A step network placed in the base circuit can be made adjustable without affecting either the extreme high- or low-frequency response.

Fig. 9 shows a step circuit connected at the input of a transistor stage having a fixed frequency-sensitive feedback loop. The current entering the base of the transistor is approximately

$$I_B = I\frac{R_{L1}}{R_{L1}+R_1}\left(\frac{1+pC_1R_1}{1+pC_1\dfrac{R_{L1}R_1}{R_{L1}+R_1}}\right).$$

The asymptotic plot of the transfer impedance of the input network combined with the fixed response of the feedback stage is shown in Fig. 9. It will be observed that there is no change in the high-frequency amplification or the extreme low frequencies as the turnover frequency is moved over the desired range.

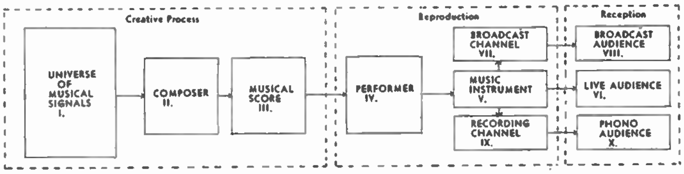
Musical Transfer Functions and Processed Music*

ANDREW G. PIKLER†

Summary—**Increasing complexity in the combinations of music and electronic technology suggests a general model of the musical transfer function. Conventional transfer includes: 1) universe of tones, 2) composer, 3) score, 4) performer, 5) musical instrument, 6) audience. Recording, broadcast, electronic synthesis and composing music by the computer either extend or modify the conventional transfer. Under "processed music," alternating and synchroneous applications of live and mechanical music, time-processed music and the more visionary polyphonic separation are described. The method of juxtaposition described permits automatic scoring of ensemble performances.**

Musical Transfer Functions

Before outlining and generalizing the connections of music, electronics and automation, it may be well to formulate a model for the musical transfer function[1].



Survey of the Functions

I→VI	Conventional transfer function
III–IV–V	its nucleus
I–II–III	creative process
IV–V	reproduction
VII, IX	technological relay channels
VI, VIII, X	expanded audience
I	natural resources
II, III, IV, VI, VIII, X	human factors
V, VII, IX	machine components

Fig. 1—The musical transfer function.

Fig. 1 portrays the conventional form of the musical transfer function and certain channels of added audiences. Musical messages are selected from the universe of musical tones (I), *i.e.*, a finite set of sine waves associated with harmonics and laid out in the system of musical scales. Hereafter, the composer (II) selects tonal combinations to cover the various dimensions of music (timing, melody, harmony, counterpoint, dynamics, timbres, etc.) in accordance with the rules of composition. He prepares a permanent record of the composition in conventional musical notation (III). The performer creates a physical input (IV) fed into a musical instrument (V), mechanical or electronic. Next an audible message is conveyed to a live audience (VI), or to a remote (VIII) or delayed (X) audiences via broadcast (VII) or recording (IX) channels, respectively, or via certain combinations of such channels (not represented in the diagram). The legend to Fig. 1 surveys these components of the transfer in various combinations.

The conventional transfer function (I–VI) can be altered in several ways by changing the role of the human and machine components. In fact, the preelectronic age has produced several forerunners to machine music, which actually eliminated the performer and his musical instrument. The music box, barrel organ, glockenspiel, pianola are the best examples. Here the performer was replaced by various kinds of programmed inputs into machines, which took the shapes of spikes on revolving cylinders, or of perforated disks or punched tapes. Mozart composed an original piece for a mechanical organ ("Phantasie für die Orgelwalze") and Beethoven for the "Panharmonicon" ("Battle Symphony"). During the past ten years, electronic and computer technology has modified the conventional transfer function in two ways; by reproducing music and sound effects electronically by means of synthesis, and by using computers themselves to *compose* music.

Electronic synthesis [2]–[5] encompasses the three subcases sketched in Table I:

1) Reproduction of an existing musical piece in synthesized form. (RCA Music Synthesizer, Princeton; 1955.)
2) Composing original musical pieces for the tape recorder to exploit tonal and temperal patterns which could be generated only by means of electronic synthesis. (Westdeutscher Rundfunk, Cologne, 1951; Columbia Electronic Music Center, Luening and Ussachevsky, 1952.)
3) Producing sound effects (musique concrète) from a white noise. (Cologne; Paris, Centre d'Études de la Radio-Télévision Française; Munich, Telefunken.)

These comparisons of Table I and Fig. 1 reveal the respective modifications in the transfer of musical information. Electronic synthesis not only eliminates the performer and the standard musical instruments but every form of mechanical sound sources such as the reeds and bars used in the automatic music of the preelectronic age.

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† At the USN Medical Research Laboratory, Groton, Conn. The opinions expressed in this article are not necessarily the official views of the U. S. Navy.

The second and most recent modification of the musical transfer function involves the use of the electronic computer for composing music. Computer music is created in two ways. (Fig. 2) [6]–[8]. In one, musical notes are generated in random sequences in the form of binary coding; hereafter, the actual selections are made by the computer programmed in terms of the rules of composition. In the second application of the computer, musical notes are sequenced in terms of transition probabilities calculated from note-to-note combinations derived from actual musical examples. In both applications of the computer, the live composer is replaced by programming logic. The music emerges directly in binary language which can be decoded into a conventional musical score. Channels of reproduction may be available in any of the combinations discussed above. However, it appears to the writer that the actual objective of computer music is not the replacement of the live composer. From computer models are expected to come the understanding of the composition process itself, as well as statistical analyses of the work and style of particular composers.

Processed Music

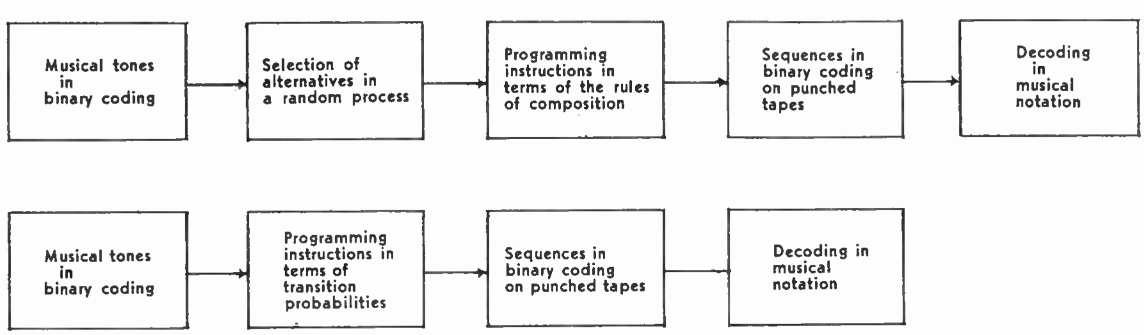
In this paper the writer wishes to discuss a fourth class of modified transfer functions for which he proposes the quite general term: Processed Music. The latter does not involve any change in the creative phases of the process. It merely represents a semiautomatic form of musical performance; namely, alternating, simultaneous and time-processed combinations of mechanical and live musical operations.

The first application of Processed Music is *successive alternation* between live and recorded music in musically perfect continuity. Such linkage is not as simple as it may seem. In September, 1960, at the Audio Fair in New York, the Fine Arts Quartet performed entire movements in preprogrammed alternations. The tape recorder, after having been started for playback, was running continuously, including the silent portions, *i.e.*, rests in the music. Naturally, there can be inherent discrepancies between the petrified time reference of the recorded sections and the flexible time scale of the live artists. This must be overcome and the live music has to be threaded into the texture of the running recorded music. The latter can be achieved only with the aid of barely audible cueings of the running silent sections of the tape, to be picked up by the leader of the live artists. Successive alternations may present some nice applications for home use.

TABLE I

Electronically Synthesized Transfer Functions

Synthesis	Signal Universe I.	Composer II.	Result of Creative Process III.	Input IV.	Instrument V.	Output VI.
1) of existing compositions	conventional	conventional	conventional	punched tape for tonal controls	harmonic generator	phonograph disk
2) of music composed originally for the tape recorder	electronically synthesizable musical patterns	composer of electronic music	a musical score with added annotations	punched tape for tonal controls	harmonic generator	magnetic tape
3) of sound effects	white noise	composer of sound effects	a script	punched tape for tonal controls	white noise generator and filters	magnetic tape



I. In the first type of logical process, alternatives, note by note, are generated at random, then screened according to the rules of composition.
II. In the second type of logical process, the computer generates sequences of the alternatives to satisfy some set of transition probabilities. ("Markoff-chains".)

Fig. 2—Composition of music by a logical computer.

The second form of processed music is *synchronization* of musical contexts. Synchronization of live music with the actual playback of mechanical music has had a familiar, though informal history since the advent of the phonograph and the radio. Many of us like to *sing* or *play with*. Before the era of long playing records, disks were produced for the purposes of "Spiel mit," *i.e.*, *add a part*. For home use rather than for performance in public, one can draw some satisfaction from the rightly chosen combinations of live and mechanized music. The performance of Mendelssohn's Octet at the Audio Fair convinced this writer that the often expressed, rigid underestimation of the play-with system is too severe. The Fine Arts Quartet, in a live performance, added their parts to the prerecorded ensemble of the first four parts and the effect of this ensemble reached the level of artistic perfection.

An even more exciting task is synchronization of recorded music with recorded music. In an article published in Europe in 1942 [9], this writer anticipated some esthetical and philosophical problems emerging from the predicted combinations of live and mechanical music. At that time one did not think in terms of man-machine systems and information processing. The writer referred to a modern philosopher (Bergson) who had depicted the conflicts of live and mechanical events, ranging from tragedy to comedy. However, in the musical sector, some normal balance between the two conflicting forces seemed to this writer to be manageable. Accordingly, he anticipated, that in all combinations of live and recorded music the human performer would have to be condemned to the humble servant's role. In other terms, only accompaniments could be added to prerecorded solo parts, while the reversed pattern would seem to be unmanageable. Nevertheless, according to the writer's consideration, an intermediate case demanded special consideration. Is synchronization of *two musically balanced parts* eventually viable? It came as a surprise to him that in 1946, Heifetz undertook the spectacular task in Bach's "Concerto for Two Violins" of synchronizing the accompanied first solo violin part and that of the second violin. This interesting, so to say historical recording [10], was based on techniques of *superposition* or *mixing* of the two parts on one sound track. Before the advent of dual channel recording, there were at least two methods to achieve mixing (Fig. 3).

At the meeting of the Division of Esthetics of the American Psychological Association in September, 1958 [11], this writer presented his own tape-recorded synchronizations prepared with a new method termed *juxtaposition* [12], [13]. He was primarily interested in improving the old method of mixing with the use of the dual channel techniques. Here the performing synchronist is relieved from the awkward task of wearing earphones and can be cued by a semi-loud or soft playback of the prerecorded first part. By separating the two parts on two tracks, one can exclude a source of distortions which is inherent in the superposition of a second track upon the first. The writer termed the new method *juxtaposition* in contrast to the previous method of mixing or *superposition*.

Table II compares the merits of the two methods of synchronization. Not only are the distortions arising from mixing avoided, but this method also permits separate loudness control on both tracks. This added flexibility can utilized in recording, playback and preparation of new mastertapes.

What are the advantages of such flexibility? The writer has "juxtaposed" his own piano accompaniment to the "Song of the Pagliacci" (Leoncavallo) sung by Caruso. The only objective was to secure a new master copy in which the old accompaniment could be eliminated psychophysically, rather than physically, *i.e.*, by means of *masking*. The flexible combination of the two tracks thus permits the best adjustment of the playback levels of the voice and of the new accompaniment. No stereophonic objective is involved here. In fact, playback and the new mastercopy should be administered via single channel.

Adding an accompaniment to an accompanied solo piece creates a somewhat different task. The writer added ("juxtaposed") Schumann's piano accompaniment to a record of Bach's "Bourrèe," a solo piece for

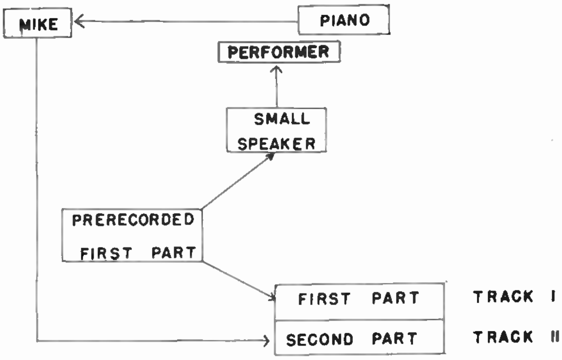


Fig. 3—Synchronization by juxtaposition on dual channel tape.

TABLE II

COMPARISON OF THE MERITS OF "SUPERPOSITION" AND "JUXTAPOSITION"

Superposition	Juxtaposition
Distortions because of mixing inevitable.	No distortions caused by mixing.
The synchronist must wear earphones or Rerecording must come from live playback and pickup through a mike.	The synchronist must not wear earphones and Direct recording possible without a mike.
Intensity relations are rigidified.	Intensity relations between the two tracks are adjustable.
Loss of stereophony.	Stereophonic and directional effects can be synthesized.
Data processing from half tracks unmanageable.	Automatic data processing from half tracks manageable.

the violin. Since there is no accompaniment in the original record, there is no task of psychophysical masking. Levels could be adjusted appropriately in the playback.

The next recording task pretains to juxtaposed synchronization of musically *balanced* parts such as in *duos* in chamber music. As an example of this type of processed music, the writer has recorded and demonstrated [1]–[11] Mozart's "Sonata for Two Pianos," both parts in his own performance. Here the objective was rigid directionality to represent the lively competition between the two pianos as opposed to the general pattern of the nondirectional kind of stereophonic liveness. In the recording it was desirable to minimize cross talk from the first track to the second track while the second track was recorded. This restriction was not too severe since the psychophysical effects of masking make small amounts of cross talk inaudible in the playback. Cueings were used on the first track to bridge over rests and these did not have to be erased inasmuch as they were inaudible in the joint playback from two tracks.

It is noted, that in the foregoing case, the directional effect is merely a result of signal processing. In a contrast to conventional stereophonic recording, the effect is not based on the natural position of sound sources in the sound field. The performer-"synchronist" is now *double* both in time and space and the resulting directional effect is *synthetic*.

Our next subject is *time-processed music*. As an example one may take Schumann's brilliant, but next-to-unplayable "Toccata" for the piano. The writer has recorded this piece at half speed in terms of the metronomic notation of the tempo and an octave lower. To secure retransposition of the musical context one octave higher, one will have to play back this recording at double speed. We prepared several other time-processed recordings of critically difficult passages of Chopin's "Etudes" for the piano [1]. These experimental recordings raise new problems:

1) What kind of musical context could be served best by means of time compression to procure satisfaction to the performer and the listener?
2) What is the rate of the esthetically acceptable time compression? This problem can be rephrased: What is the extent of the *musical interval* in the scale, within which the spectral distortions from time-expanded recording and time-compressed playback are non-noticeable to the trained listeners? Are there pyschophysical threshold values of detectability *e.g.*, for time-processed piano music?
3) What specific adjustments by the performer secure realism?

There seems to be an open field for psychophysical mapping in order to explore the psychophysical ranges of probabilities of detection with respect to *time-processed timbres*, as a function of rate of compression, frequency range and loudness.

On an introspective basis, the writer predicts that, for the piano, the rate of frequency compression of $\frac{2}{3}$, which corresponds to the musical interval of a Perfect Fifth, is esthetically acceptable. This would mean that pianists gifted with the $\frac{2}{3}$ fraction of Wladimir Horowitz's efficiency in the speed of muscular functions, could still produce good Schumann "Toccata" recordings. For the recording of unplayable pieces, the RCA music synthesizer discarded the performer altogether. When resorting to the techniques of time processing, one can still employ live musicians, whose efficiency is somewhat short of the physiological limit.

For the sake of completion, one must recall the possibility of a second approach, *i.e.*, the use of the now standardized Fairbanks-apparatus serving the audiological study of time-processed speech [14]. Its principle is different from the above in that it is based on sampling from the complex wave. The resulting loss of acoustical information seems to be more critical in those dimensions which concern music, as compared with the above-mentioned procedure of time-processing. Comparison of the two systems from the point of view of musical fidelity is open for research.

A last proposal concerning Processed Music is admittedly visionary. A new operation could be thought of to decompose and *desynchronize* polyphonic musical contexts which were recorded or broadcast. Example: Can a four parts counterpoint be decomposed with the intervention of the human operator such that the four parts be led into four separate channels? The writer raised this problem in 1942 [9], then in 1955 [12] and is still not closer to the solution. He would expect that a keyboard instrument activating a set of band-pass filters, handled by the human operator in the process of *auditory pursuit tracking* [19], could render such a task possible. The envisioned instrument, the *Polyphonic Separator*, represents a challenging problem for electronic technology, for the theory of information and communications and may have even practical applications in signal detection.

Display of Transfer Functions and Music Psychology

A limiting case of synchronization by juxtaposition deserves special considerations. For all practical purposes, the foregoing techniques permit total separation of the live input from the prerecorded information on two tracks. Juxtaposed recordings of such type have an interesting application in experimental music psychology in that they afford *cross correlation analysis* of the two information flows by means of *automatic data processing*. The proposed approach would revitalize the performance scores of Seashore and associates [15] in a

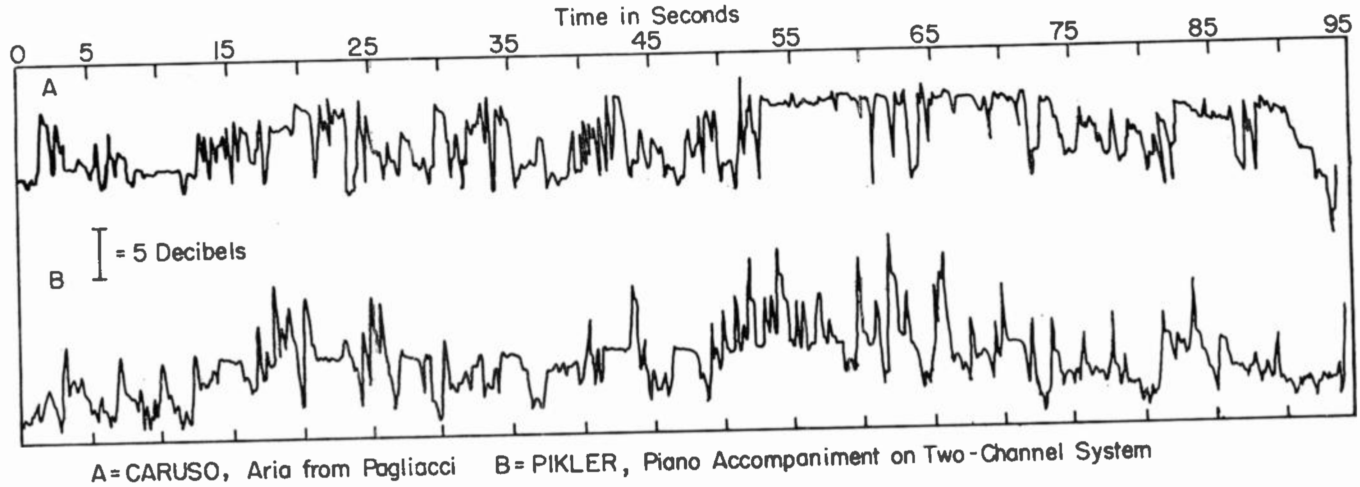


Fig. 4—Time-intensity spectrum in the ensemble of a duo.

new dimension. About 25 years ago, using the techniques of phono-photography, the latter recorded short samples of *solo* performances and resorted to rather laborious, semi-automatic methods for plotting and analyzing such performance scores, note by note. On the other hand, the new technique of synchronization on dual tapes permits continuous and automatic evaluation of the *precision of ensemble music* [16], primarily in a duo. *i.e.*, the simplest form of an ensemble. Here, evaluation is essentially *statistical* as opposed to the classical approach of Seashore, that, like the teacher of music, assesses the performance *note by note*.

For the suggested task, one will have to use a very soft directional speaker for cueing, and an appropriately placed directional mike in order to prevent intrusions of the prerecorded part into the juxtaposed second track. The writer's Caruso recording was actually made with such an objective in mind. Since he had no access to automatic data processing, he availed himself of a conventional pen recorder of voltages and hereby confined his analysis to the display of the time-intensity spectra on the two parallel tracks (Fig. 4). By collecting time samples, it was possible to assess the *statistical average of the time lag*. In fact, the latter represents the basic datum of ensemble synchrony. It appears that a good ensemble musician works with a 30-msec statistical lag, which would be sufficient to create the ensemble illusion. From such pen tracings it is also possible to evaluate the discrepancies between the slopes of the crescendi and decrescendi as performed by the soloist and the accompanist, respectively. On the other hand, such performance scores are uninformative concerning the absolute levels and the level differences of solo parts and accompaniments. It will be recalled that, in the phase of the recording, these levels were handled quite arbitrarily.

For the pickup of the separate parts of a live ensemble, contact microphones like those used inside electric guitars may be attached to the resonating chambers of the instruments, (*e.g.*, violins). A live ensemble of string instruments equipped with similar microphones, performed at the Acoustical Society's May, 1947 meeting. Such a separation technique with multichannel tape recording might now be easily applied for scoring polyphonic performances, as for example, the intonational standards (Pythagorean, Diatonic, etc.) [17].

Another look at Fig. 4 reveals the outlines of a pattern of *pursuit tracking*. The accompanist "tracks" the soloist and the solo part constitutes the "program" for such tracking task. From such consideration in 1955, the writer arrived at the philosophy of *audio-tracking* [12] and described its diverse applications in the fields of automatic audiometry [18] and human engineering [19], [20].

Music psychology could profit along more general lines from the rationale of auditory tracking. How is synchrony generated in live musical ensembles, *i.e.*, in *men-to-men systems?* How is it possible to deceive an audience with the ensemble illusion? Performances in ensemble music can be rationalized as a *complex form of auditory pursuit tracking* in the three dimensions of time scale, pitch and loudness simultaneously. In the various forms of musical ensembles, different tracking patterns and aids are used. Figs. 5 and 6 illustrate the fundamentals of the proposed psychological theory of live ensemble music which, probably, represents the asymptote for precision of all human performances in cooperative groups.

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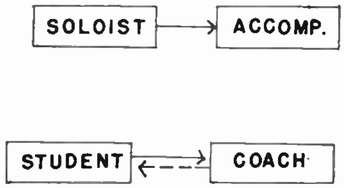

Fig. 5—Synchrony and tracking in live ensembles. (Accompaniment).

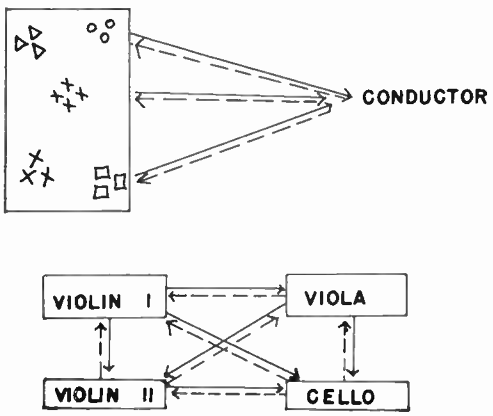

Fig. 6—Synchrony and tracking in live ensembles. (Chamber music and orchestra).

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Automatic Measurement of Phonograph Reproducers*

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Summary—**A stereophonic test record has been designed for measurement of pickup response on a curve recorder. The principles of stereophonic disk recording which have governed the design of this record, are reviewed. The new test record contains two sweep-frequency bands for the left and right channels, the signal frequency varying logarithmically with time at a rate of 1 decade each 24 seconds covering a range of 40–20,000 cps. Left and right spot-frequency tones with voice announcements have also been provided covering a range of 20–20,000 cps. Automatic recording of pickup response has been estimated to save ½ to ⅓ of manpower used in pickup and phonograph development. For automatic measurement of preamplifier characteristics, an RC RIAA generating network has been designed. A circuit for automatic starting of recorder by keying tones on the record is described.**

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Among the latest of manual procedures to yield to automation is the technology of measuring phonograph reproducer characteristics. This has come about as a result of development of a new CBS Laboratories Type STR 100 Stereophonic Frequency Test Record which is adapted for use with General Radio Type 1521-A Recorder.

A stereophonic record contains two related program channels which are identified with orthogonal modulations of the walls of a single groove. The left channel corresponds to the "inner" groove wall, the one closest to the center, and the right channel to the "outer" groove wall (away from the center). The positive directions of these modulations are at 45 degrees and −45 degrees to the record surface, respectively (Fig. 1). A phonograph pickup designed to play this type of record has a single stylus acting on a pair of transducers, such as magnetic circuits or piezoelectric elements. These are arranged in an orthogonal fashion and intended to generate independent voltages when driven by the respective groove wall modulations (Fig. 2). The voltages e_l and e_r generated in the transducers drive two amplifiers and loudspeakers to reproduce the original left and right channels.

The only feasible way of testing the performance of a phonograph pickup is by using a frequency test record. Typically such records contain tones recorded at various frequencies. Stereophonic frequency test records have separate recordings for the left and the right channels. The response vs frequency of any given pickup channel produced by the corresponding record channel is known as "response-frequency characteristic" or simply "response" of the channel on the particular record. The response vs frequency stemming from the opposite channel is known as "cross-talk-frequency characteristic" or simply "cross talk." The customary units for both characteristics are db re 1 volt rms. The difference at any one frequency (or average over a group of frequencies) between response and cross talk is known as "channel separation," expressed in db.

The STR 100 record is characterized by having two sweep-frequency bands for the left and right channels, the frequencies of which vary logarithmically with time, at a rate of 1 decade each 24 sec. This correspond to a sweep rate of the General Radio Type 1521-A Recorder and Frequency Response paper at thirty $\frac{1}{4}$-in per minute speed. The frequency range covered by the STR 100 record is 40 to 20,000. The sweep band starts with a 1000-cps tone of sufficient duration to set the recorder pen to the 40-cps ordinate and to adjust the recording level to a convenient value. Upon cessation of the 1000-cps tone the frequency drops immediately to 40 cps and then rises continuously to 20,000 cps in 24 log (20,000/40) = 64.8 sec. Next, an interval is allowed for the operator to reset the recorder chart back to the 40-cps ordinate. Then, the right channel 1-kc tone is heard, being followed by the right-channel sweep. If one channel of a pickup under test is connected to the recorder (through a suitable amplifier) then the response-frequency and the cross-talk-frequency characteristics for the particular channel will be successively recorded. A typical set of curves for a high grade magnetic phonograph pickup is shown in Fig. 3. Two such sets, one for each channel are required to describe the performance of the pickup.

In previously available test records the test frequencies are in the form of spot-frequency tones not adapted for automatic recording. The STR 100 record also has such fixed-frequencies tones for the left and right channels, and unlike with the older records, each tone is preceded by a voice announcement of frequency. In this manner there is no doubt as to which tone is being played. The results of the "spot-frequency" tests are similar to those obtained with the sweep-frequency bands but the process is far more laborious, and the information between the spot-frequency tones is not revealed. Many persons who thought that they had a "flat system" when measured with spot-frequency bands

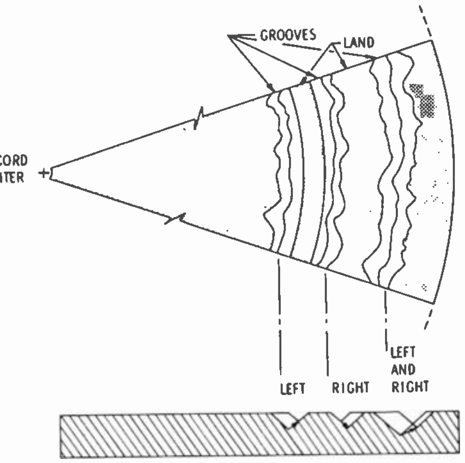

Fig. 1—"Pie" representation of a stereophonic record portraying modulation associated with left channel, right channel and combined left and right channels.

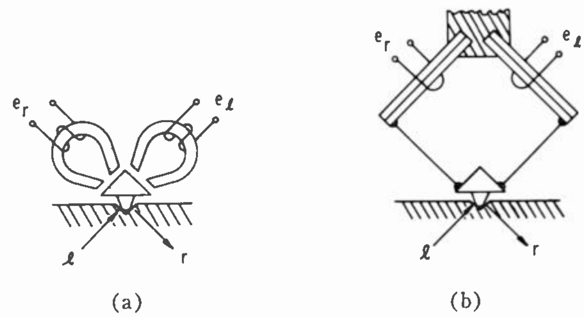

Fig. 2—Typical arrangements for stereophonic pickups. (a) Magnetic. (b) Piezoelectric.

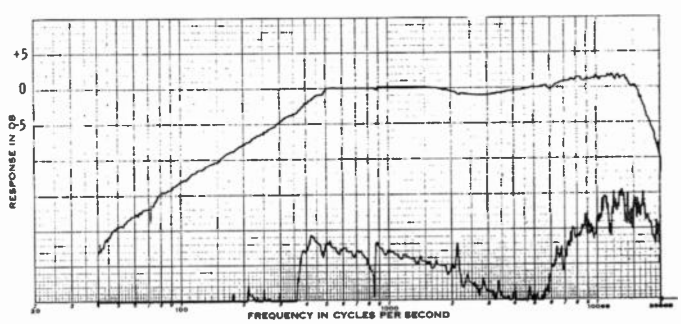

Fig. 3—Automatically plotted response of typical high grade magnetic pickup.

found the existence of resonant peaks or dips in between the spot frequencies!!

The spot-frequency bands in the STR 100 record above 500 cps were cut with the same recorder setup as were the sweep-tone bands. This permitted an absolute calibration of the sweep-tone bands by measuring the spot-frequency bands with microscope and by diffraction-of-light patterns.[1]

Economics of Automation

The economic value of automation in testing phonograph reproducers is so startling as to deserve a special mention. The development of a phonograph pickup or complete player involves considerable experimental work in which the device under test is modified by successive rearrangement or modification of components until the desired performance is obtained. Each modification is followed by response and separation tests. With the spot-frequency tests the process of obtaining these characteristics requires a better part of one hour. It has been estimated that $\frac{1}{2}$ to $\frac{1}{3}$ of manpower used in pickup and phonograph development is expended in this tedious endeavor. Delegating these measurements to an automatic device which does not mind tedium and makes no error releases trained manpower for more creative tasks and saves thousands of dollars annually. In one instance, changing from manual to automatic recording has been shown to save the cost of a recorder in a single month, and the cost of the new record in a single test.

Test Record Characteristics

As indicated above, the reproducing characteristics of a pickup or system under test are referred to the characteristics of a particular test record. These may be defined in terms of displacement or velocity of groove modulation.

Referring to Fig. 4, let one groove wall be modulated sinusoidally with a maximum amplitude y max. As the groove moves lengthwise with a velocity v it will impart to the stylus a normal sinusoidal displacement y,

$$y = y_{\max} \sin 2\pi ft, \qquad (1)$$

where f is the modulation frequency.

Velocity y' is the time rate of displacement, or,

$$y' = 2\pi f y_{\max} \cos 2\pi ft = y_{\max}' \cos 2\pi ft. \qquad (2)$$

From (2) $y'_{\max} = 2\pi f y_{\max}$ and since the rms value of a sine or cosine function is obtained by dividing the maximum value by $\sqrt{2}$, it is equally correct to say $y'_{\text{rms}} = 2\pi f y_{\text{rms}}$. Therefore, for a given amplitude of displacement y, velocity increases linearly with the frequency. If a frequency record is recorded with constant displacement then a displacement-responsive (piezoelectric) pickup will produce a constant output at all frequencies, while a velocity-responsive pickup (magnetic, moving coil) will produce an output directly proportional to frequency. On the other hand, a constant velocity recording will produce a constant output with a velocity-responsive pickup, and an output inversely proportional to frequency with a displacement-responsive pickup.

The STR 100 record has a constant displacement modulation up to 500 cps and constant velocity modulation above 500 cps. This explains the response of a magnetic pickup in Fig. 3, which is a rising straight line in db vs log frequency at 6 db per octave below 500 cps and constant above 500 cps. The deviation of response from two straight lines denotes the departure of the pickup from ideal performance. By constrast, in Fig. 3 the cross-talk curve which is 20 to 30 below the principal channel output typifies the order of mangitude of channel separation that may be expected in present-day high grade pickups.

Similar measurement can readily be performed on displacement-responsive pickups by differentiating the generated voltage. This is done by connecting a resistance which is small compared to the capacitive reactance of the pickup across the pickup terminals. Usually a 10,000-ohm resistor will suffice. The response-frequency characteristic of an ideal piezoelectric pickup terminated in this manner is similar to that of a magnetic pickup.

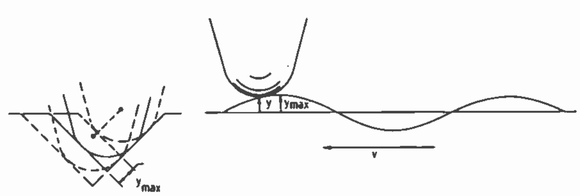

Fig. 4—Geometry of single-channel stereophonic groove.

Testing Pickup Preamplifiers

Modern $33\frac{1}{3}$-rpm records are recorded with a frequency characteristic which is a composite of several constant displacement and constant velocity segments as follows:

[1] IRE Standards Committee, "Standards on recording and reproducing: Method of calibration of mechanically-recorded lateral frequency records," Proc. IRE, vol. 46, pp. 1940–1946; December, 1958.

Up to 50 cps—constant velocity
From 50 to 500 cps—constant displacement
From 500 to 2120 cps—constant velocity
Above 2120 cps—constant displacement.

The transitions between these segments are not sharply defined, but instead they are blended together, in a manner defined by a standard of the Record Industry Association of America (RIAA).[2] The type of velocity (db) vs frequency characteristic which is obtained is shown in the graph of Fig. 5.

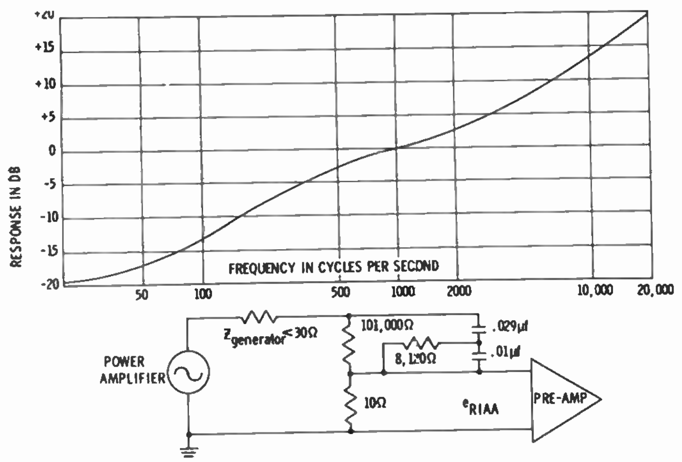

Fig. 5—RIAA recording characteristic and RC RIAA network.

Having determined that a pickup is truly a velocity-responsive device by the use of STR 100 record, it would follow that the pickup preamplifier should have an inverse RIAA characteristic. This characteristic of the preamplifier is most conveniently verified by providing an RIAA generator network between the oscillator and the amplifier under test. One such network is shown in Fig. 5 and by inserting it between the oscillator and the preamplifier, the performance of the latter can readily be ascertained. If the resultant response is uniform with frequency, then the preamplifier is properly designed for reproducing records with a magnetic pickup. A Type 1521-A recorder and Type 1304-B oscillator can be advantageously used in this test.

Over-all Measurements of Response Characteristics

The STR record and the Type 1521 recorder may be used to measure the over-all response-frequency characteristics of a playback system also regardless of the type of pickup or amplifier employed. The ideal response-frequency characteristic of a properly adjusted reproduction system, playing an STR 100 record simply will be the difference between the response frequency characteristic of the record and the RIAA characteristic shown in Fig. 5. Subtracting these two results in the following set of values:

Ideal System Response—RIAA Equalized

Frequency	db	Frequency	db
1000	0	800	+0.7
20,000	−19.5	600	+1.8
18,000	−18.8	500	+2.6
16,000	−17.7	400	+1.9
14,000	−16.6	300	+1.1
12,000	−15.3	200	+0.2
10,000	−13.7	150	−0.6
8000	−11.9	100	−0.9
6000	− 9.6	80	−1.3
5000	− 8.2	60	−2.3
4000	− 6.6	50	−3.0
3000	− 4.8	40	−4.2
2000	− 2.6	30	−5.8
1500	− 1.5	25	−7.0
1000	0	20	−8.6

Tone Arm Resonance Test

To test resonance of tone arms, loudspeakers, etc., the STR 100 record provides sweep-tone bands, left and right, from 200 to 10 cps. These are synchronized also to the Type 1521 recorder. The recorder must be set to operate in reverse, as the sweep tone begins at 200 cps and the frequency decreases during the glide.

Automatic Start Circuit

The 1000-cps tones at the beginning of each glide are intended to serve not only for level adjustment, but also for keying an automatic start circuit for the recorder. This circuit developed by A. Schwartz and A. Gust of CBS Laboratories is shown in Fig. 6 (next page).

The 1000-cycle keying tone preceding the sweep initiates the cycle. All relays are initially deenergized as shown in the schematic diagram. Left- and right-channel inputs are combined in the cathode of V1 insuring that the keying tone will be present for either direct or cross-talk measurements. The cathode follower output is fed to the V2 high gain amplifier through a high Q L-C 1000-cps filter allowing only 1000 cps to feed through. Following this stage is a cathode follower V3 employed as a power amplified to drive a sensitive relay K1 (Elgin Advance) after rectification by the two IN2482 diodes. A Zener diode and clipping range control prevent high signal levels from overheating the sensitive relay.

With K1 energized, relay K2 is energized and locks itself across the power supply through contact 1. At the cessation of the keying tone, K1 is deenergized—thereby actuating K3 which starts the recorder motor at the instant the sweep begins. At the termination of the sweep band, the reset switch is manually set at RESET momentarily to deenergize K2 and the circuit is the ready for the next sweep.

[2] "Standard Recording and Reproducing Characteristic," Record Industries Association of America, Inc., 1 East 57 St., New York 22, N. Y.

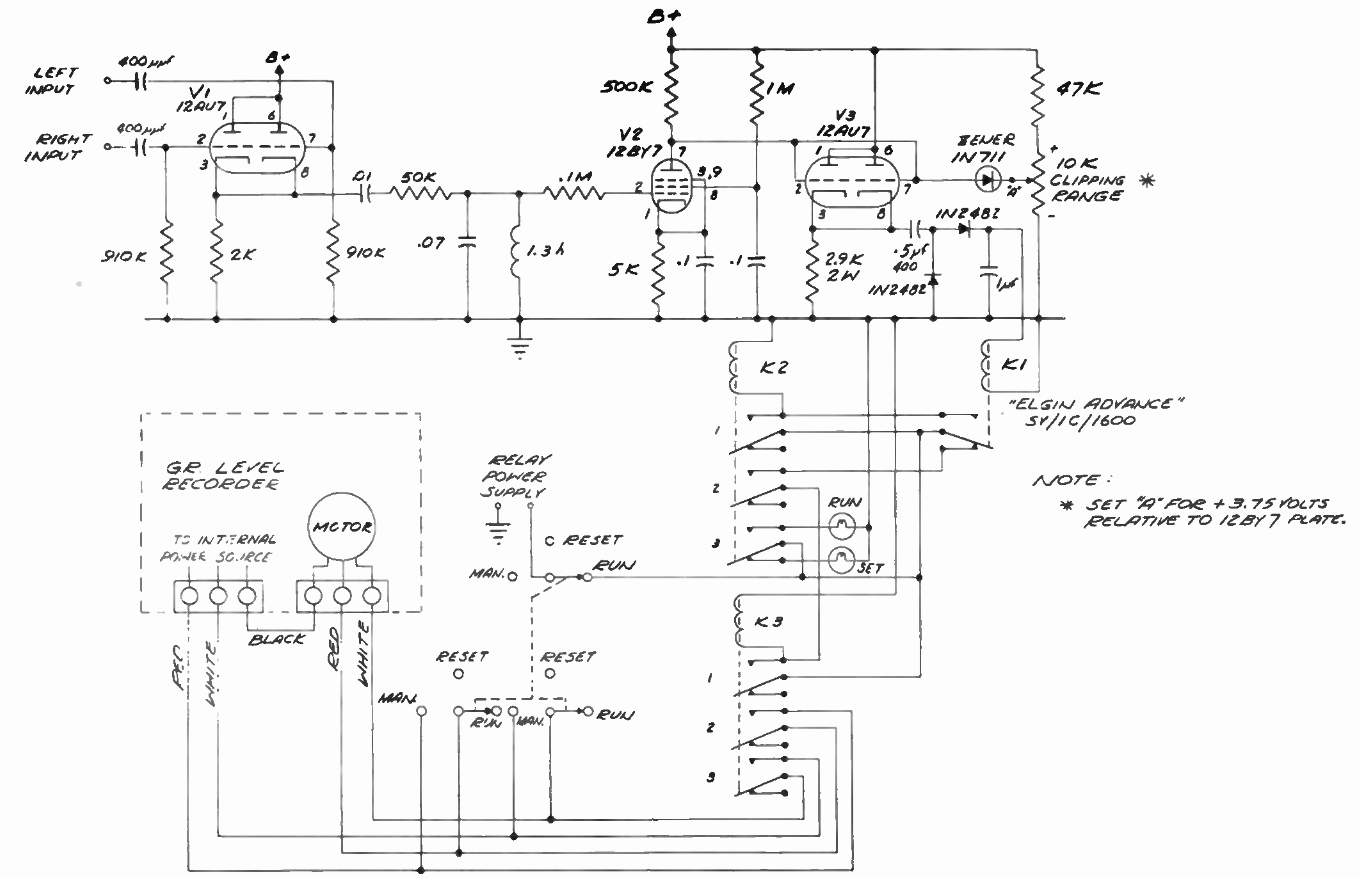

Fig. 6—Synchronizing circuit for automatic starting of recorder.

Correspondence

Enhanced Stereo

Comments on Letter From H. K. McKechnie*[1]

The use of a single microphone per channel in pickup of sound for broadcast and recording is made possible by placement of the microphones sufficiently far from all performers so that there is not a level variation from performer to performer dependent upon his position with respect to the microphone. If the microphones are placed on a stage in near proximity of the performers, then there is an intensity variation due to the variable distance from performer to microphone which is as noticeable as the phase difference due to the use of multiple microphones. Good recording techniques therefore use single microphones per channel placed at a relatively large distance in front of the performers. Two or more such channels result in good reproduction and is truly stereo, but the difference between good quality single channel and good quality multiple channel is not sufficient to satisfy the stereo enthusiast. I personally do not advocate "enhanced stereo," but wish to point out techniques which a good portion of the recording, as well as the broadcast industry, are using in order to give the consumer something which he can say is different.

It is necessary to point out that in sound reenforcement systems it is often required that multiple microphones per channel be used. The inherent gain in a live system without feedback is limited by the sound absorption in the room, for normal placement of loudspeakers. The levels provided by the reenforcement system then become directly related to the distance from performer

* Received by the PGA, December 26, 1961.

[1] H. K. MacKechnie, IRE TRANS. ON AUDIO, vol. AU-9, p. 180; September–October, 1961.

to microphone. Very satisfactory systems have been designed and are in use, using multiple microphones per channel, however the number of microphones and their placement become extremely critical.

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Vanderbilt University
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Additional Comments[2]

Some 10 years ago I asked B. B. Bauer whether he had some advice on microphone placement for stereo recordings. He suggested that we conduct our own experiments and share the results with interested parties, which the Cleveland PGA Chapter promptly started and has been doing since. Besides several stereo symposia, we have worked with George Szell and the Cleveland Orchestra, and I spent a year in France recording some 50 concerts, operas, and other musical events with a three-channel system.

The Editor of IRE TRANSACTIONS on AUDIO has finally stirred up the hornet's nest of opinions concerning microphone techniques and I now feel audacious enough to add my own, for whatever it may be worth.

The use of multiple microphones for stereo as described by Benson,[3] unfortunately is not quite as obsolete as MacKechnie[1] and myself would like to think. During my stay in France, I was privileged to experiment along with the engineering unit of the Radio-Diffusion-Télévision Française, the government broadcasting agency. While all my recordings were taken with only one microphone for each of the three channels—as MacKechnie and many outstanding Tonmeisters recommend—the French engineers insisted on as many as five mikes per channel, with some of them bridging both sides to reduce the "center-hole" effect. Even though they were quite impressed by the spaciousness and clarity of the single-mike recordings, it took a drastic demonstration to undermine their faith in multiple stereo pickup. We had recorded a performance of Berlioz' Requiem at the Théâtre des Champs-Elyseés for playback to the press at the Cinerama Theater of one, two, three and four channel systems. During the set-up tests with the powerful speakers behind the Cinerama screen, I deliberately switched left and right channels of a recording made with many microphones. No one raised a voice of objection. When I mentioned to the Tonmeister that he was listening to a mirror image of his own tape, he thought it had sounded strange, so we straightened things out. By now all the personnel present had caught on to the experiment and were arguing among themselves which was the correct and which the reverse image. Some insisted that the channels must have been mis-labeled when, in fact, the use of multiple microphones had befogged the panorama. The appearance in the score of fanfares placed in the corners of the concert hall finally settled the argument.

I have since heard a few excellent multi-mike recordings—notably by Deutsche Gramophone who also use selective octave amplifiers, another anathema to the purist—and many more outstanding "single" microphone pickups. Panels of musicians, including conductors such as Dr. Szell and composers Stravinsky and Barber, have expressed their preference for the widely-spaced one-mike-per-channel technique while auditioning our own tapes in a typical home environment.

The best results were obtained with three omnidirectional condenser microphones spaced eleven feet apart and suspended about twenty feet above stage level over the first row of the audience. This set-up provides sufficient reverberated signal to convey information on the concert hall acoustics without deteriorating the direct sound from the orchestra. Since a few inches in height make a great deal of difference if the stage is backed by a shell, the exact distances will vary from hall to hall, but, once established, the results may be repeated without difficulty.

For concerti the center mike was lowered to about ten feet above the stage, or a separate unidirectional microphone located several feet in front of the soloist was mixed in with the regular center channel. Using two pickups for the middle did not distort the panorama since they were both on axis. It did permit "spot lighting" the single artist without either overemphasizing the center portion of the orchestra or changing the ratio of direct to reverberated sound and with it the apparent distance to the soloist.

After compiling the listeners' reactions to these experiments I am as strong an advocate of "trinaural" as MacKechnie and am as convinced as he that "nothing less will ultimately be satisfactory."

HERBERT H. HELLER
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Cleveland, Ohio

[2] Received by the PGA, January 15, 1962.

[3] R. W. Benson, IRE TRANS. ON AUDIO, vol. AU-9, pp. 63–65; May–June, 1961.

Contributors

Hanswerner Pieplow (SM'57) was born in Warmbrunn, Germany, on April 29, 1911. He received the Diploma-Engineer degree in 1933 from the Technical University, Berlin.

He joined the Research Institute of the AEG in Berlin where he was engaged in the development of the first thyratron controlled welding machines. He was also associated with the development of high-speed cathode-ray oscilloscopes and on aircraft altimeters. Since the war he has been active in studio and audio techniques. He joined *Grundig Radio Werke*, Furth, West Germany, in 1954 and became Head of the tape recorder division in 1957.

Mr. Pieplow is a member of the Society of German Engineers (VDE) and of the German Standards Committee for Magnetic Recording.

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Herman H. Scott (M'35–SM'43–F'52) was born in Somerville, Mass., on March 28, 1909. He received the B.S. and M.S. degrees in electrical engineering from the Massachusetts Institute of Technology, Cambridge, Mass., in 1930 and 1931, respectively.

His early experience includes work with the Bell Telephone Laboratories, New York, N. Y., on sound motion pictures and high quality broadcast systems from 1929–1930. From 1931 to 1946 he was associated with the General Radio Company, Cambridge, Mass., latterly as Executive Engineer in Charge of Audio, Acoustic, Broadcast and related developments. From 1947 to date he has been President of H. H. Scott, Inc., Maynard, Mass., manufacturers of electronic instruments, high-fidelity components, etc. He has been a special Lecturer at the Amos Tuck School of Business Administration of Dartmouth College, Hanover, N. H.

He is the inventor of the RC Oscillator, R-C Selective Circuit, and various RC filters, as well as early electronic sweep circuits, pulse counting frequency meters, etc. He is also inventor of the Dynamic Noise Suppressor and various types of electronic filters and, more recently, of various stereophonic systems and devices.

Mr. Scott is a Fellow of the Acoustical Society of America and the Audio Engineering Society. In 1951 he was presented the John H. Potts Memorial Award by the Audio Engineering Society "for outstanding achievement in the field of audio engineering." He is a Past Chairman of the Boston Section of the IRE and has served on various IRE committees including the Board of Editors. He is President of the Audio Engineering Society and Past Chairman of the Board of the Institute of High Fidelity Manufacturers.

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Daniel R. von Recklinghausen (S'49–A'54–M'57–SM'60) was born in New York, N. Y., on January 22, 1925. He received the S.B. degree in electrical engineering in 1951 from the Massachusetts Institute of Technology, Cambridge, Mass.

Previous to receiving his degree, he was employed at Rohde & Schwarz, Munich, Germany and the Research Laboratory of Electronics, the High Voltage and Acoustics Laboratories of M.I.T. doing development work on UHF and SHF meters and generators, reverberation devices and recording and studio facilities. He joined H. H. Scott, Inc., Maynard, Mass., in 1951 as Project Engineer responsible for the development of sound analyzers, tuners, amplifiers and other acoustical instruments. In 1955 he became Chief Research Engineer at H. H. Scott, a position he holds at present.

Mr. von Recklinghausen is Chairman of the Tuner Standards Committee of the Institute of High Fidelity Manufacturers and a member of the IHFM Amplifier Standards Committee. He has been a member of Panels 4 and 5 of the National Stereophonic Radio Committee and was Chairman of Subcommittees 4.1 and 5.4, NSRC. He is a member of Tau Beta Pi, Sigma Xi, Eta Kappa Nu, and the Audio Engineering Society.

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Harold Fristoe (S'41–M'47–SM'50) was born in Windsor, Mo., on June 4, 1909. He received the B.S. degree in 1933 from Eastern Michigan University (formerly Michigan State Normal College), Ypsilanti, the B.S.E.E. degree in 1941 from the University of Arkansas, Fayetteville, the M.S.E.E. and E.E. degrees in 1947 from Oklahoma State University, Stillwater, and the University of Arkansas, respectively, and the Ph.D. degree in 1953.

In 1933 he joined the faculty of John Brown University, Siloam Springs, Ark., and in 1935 established the Radio School. In 1941 he became a member of the Department of Electrical Engineering, Oklahoma State University. At present, as Professor of Electrical Engineering, he is teaching in the field of electronic devices and electronic circuit theory and applications. He has spent the past five summers with Texas Instruments Inc., Dallas, developing applications for solid-state devices.

Dr. Fristoe is a Registered Professional Engineer (Oklahoma) and is a member of OSPE, NSPE, ASEE, AES, Eta Kappa Nu, Sigma Tau, and Pi Mu Epsilon.

❖

Andrew G. Pikler was born in Budapest, Hungary, on June 16, 1907. He studied social sciences, philosophy and law at the Universities of Budapest, Berlin and Paris and received the Ph.D. degree in 1929.

He was invited to the United States on a postdoctoral program in mathematical economics in 1946.

Since 1951 he has been a Research Psychophysicist with the U. S. Navy, and at present is at the USN Medical Research Laboratory, USN Submarine Base, New London, Conn. His research interest is centered on the problems of musical hearing, absolute and relative pitch, the theory of auditory tracking and its applications in human engineering. He has an extensive background in music as a pianist and has published music in the U. S. He has published work on the kinetics of monetary circulation, the applications of the entropy principle and of the equations of field physics for models in mathematical economics and psychology.

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Benjamin B. Bauer (S'37–A'39–SM'44–F'53) for a photograph and biography, please see page 181 of the September–October, 1961, issue of these TRANSACTIONS.